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PROJECT RECORDING & SOUND TECHNIQUES VOLUME 11, ISSUE 4 APRIL 2000



ON THE COVER: Danny Saber strikes a

pose. Photo by Edward Colver.



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EQ (ISSN 1050-7868) is published monthly plus Buyer's Guide in December by Miller Freeman PSN Inc., 460 Park Ave. south, 9th fl., New York, NY 10016-7315. Periodicals postage paid at New York, NY and additional mailing offices. POSTMASTER: Send address changes to EQ, P.O. Box 0532, Boldwin, NY 11510-0532. SUBSCRIPTIONS; U.S. \$29.95 for 1 yr. (13 issues); CANADA add \$10.00 per year for surface; other countries add \$15.00 per yr. for surface; All add \$30.00 per yr. for Airmail. All subscriptions outside the U.S. must be pre-paid in U.S. funds-by International Money Order, checks draw from a bank located in the USA Viso, Master Card ar American Express. Back-issues \$5. Printed in the U.S.A.

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TRUE BREW

I was just reading your section on performing engineers [Feburary 2000] and came across the article by Peter Gorges. In general, I have learned a great deal from the pages of your magazine. Unfortunately, there was one error I picked up on. Koelsch beer is actually an ale — not a lager. I have made several award-winning ones myself in the past. While music production is my parttime job, homebrewing is my serious hobby. (After all, you have to have something to offer to your clients while they work, right?) Just thought I'd let you know...

> Chuck Hanning SoundBase Studio Owner/Engineer 1998 Delaware Valley Homebrewer of the Year Malvern, PA

CURSES!

I just wanted to get my two cents in regarding the ongoing debate about printing swear words in EQ. While there's probably a limit to how much swearing could be considered offensive to the "general public," I believe there's nothing wrong at all with printing the occasional swear word. If it happens to represent the way a given person thinks and feels and expresses him/herself, then so be it. Like it or not, swearing is one more tool we use to express ourselves linguistically - and who are we as artists/creators to censor someone else's conscious choice to use that tool? And, in any case, I can't honestly see how "f*ck" is less offensive than, er, the, ah, F-word.

Our society suffers from enough censorship in many subtle and not-sosubtle ways. I, for one, couldn't care less if you print swear words.

> Daniel Griffin Madrid, Spain

LET'S MAKE A DEAL

I liked Steve La Cerra's editorial in the Jan. 2000 issue. I think he made some valid points. I also thought that the responses to the editorial made valid points, too. It brought to mind a real-life experience. (Names have been changed to protect the innocent and guilty.)

Around the time the Alesis Midiverb II came out, the band I was in was thinking of getting one. I volunteered to price shop and make a recommendation. Deciding to give local retailers first shot, I went to a store near me to see what they were selling the unit for. Mick, the manager/owner, was not in, but the assistant helped me out. Judy was not sure what they were selling the Alesis for since it was so new and they did not have one in store. She got out the "dealer price guide," which showed the price to be \$325. Judy said they would probably sell the unit for \$295, if I remember right. I thanked her for the info and left the store, got into my car and leafed over to the Alesis ad in the current issue of a well-known pro magazine. There, in the ad, was the price: \$285 suggested retail.

Now this store is one of those that believe it's worth paying a bit more because they would be there to help if there were any problems. I don't think paying more than advertised suggested retail is quite the way it should be done. I guess that if you can find a good local store with folks who know their stuff, in a certain respect, you ought to buy from them. But I do understand the financial concerns that enter in. When dealing with gear that is practically disposable like that \$99 reverb - going for best price is not all that bad. When something like that breaks and you could buy a new one for a few dollars more than it costs to fix it, store help really doesn't matter much.

And, let's face it, sometimes you just can't have everything you want, when you want it. If a piece of gear is worth having, it's probably worth saving for.

Thanks for letting me put in my half-penny's worth.

Ron Carlson Boone, IA

PRESSING MATTERS

Reading Roger Nichols's March column on his CD pressing problem, I conclude the test pressing was marginal in some way so it would play on the better machines and not on cheaper ones, at least without having a very high error rate. That is what he was hearing — not some mystery jitter factor. (This is a chronic problem with 80-minute CD-Rs).

When he compared the files from the reference CD and the test pressing, he was using the CD player in the computer, which could deal with the marginal CD.

What he needed to do was measure the errors on the test pressing using a cheap CD player. How can this be done? I recorded an inaudible low-level pulse at the beginning and ending of the CD. Then I played the CD audio digitally into the computer using a separate CD player. After trimming the resultant soundfile to match the original, I used the DOS "COMP" command to compare the soundfiles. They were identical. If the CD player doesn't have a digital output, then it is more difficult. You have to be able to match the amplitudes and take the difference.

I am sure if you did a comparison test with his test pressing played on a portable CD player, there would be major errors.

> David Hadaway via Internet

Roger Nichols responds: As I stated in my article, there were no errors on the pressed disk. All of the CD players used, the \$100 portable thru the \$10,000 Oxygen-free unit, fed the same converters through digital outputs. I played it back on a StageTech CD Error checker at the same time it was logging Blers. The average error rate was 10 to 20, CD plants bounce discs at 320, and interpolation does not kick in until 3000. No error-induced audio problems here. I did make one pass with digital audio in to the computer for comparison. No errors. No matter how accurately you adjust the analog out of a CD player, and especially without a CD player that accepts external sync, you will never get a result file of all zeros, even if you play the same CD back off the same player twice in a row. Bottom line. After testing all of the CDs by Warner Bros. Records, WEA CD pressing Plant, Sony Studios, Classic Sound Mastering, and myself, there were no significant errors on any of the CDs.]

CORRECTIONS

The source for Taiyo Yuden CD-Rs David Torrey mentioned in the March issue's "CD-R Checklist," Marilyn Sierchio, has moved. Reach her at Total Media, Inc., 800-776 8273.

In the Oversight of the Century, we neglected to credit technology editor Craig Anderton with the copious amount of research and endless legwork he put into March's Winter NAMM 2000 Wrap-Up. A thousand apologies, Craig!

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LOOKING FOR LINUX

Q I wanted to find out about using Linux for music production. Are there any good Web sites to get more information? Thank you.

> Jeff via Internet

Linux has a long way to go until it will be used the way MacOS and Windows are for music production. However, it is currently useful for conversion between formats, encoding to MP3, and serving content up to the world as a server. For now, you're best off using a Windows or MacOS (or even plain old analog) for the meat of your production and then moving to Linux (or FreeBSD) when you want to get it out to the Internet. As for Web sites, www.freeamp.com provides Linux support for MP3 players and Xing (www. xingtech.com) has an MP3 encoder. Streaming servers available for Linux include RealNetworks (www.real.com) and the MP3 streaming solutions Shoutcast (www.shoutcast.com) and Icecast (www.icecast.com).

> FezGuys EQ Columnists

CONSOLE CONUNDRUM

I need a dedicated mixing board for mixing acoustic drums. The people I work with are collectively looking at a mixing board tasked specifically for mixing my kit and are looking very hard at a 16-channel, 4-bus unit that features 4 parametric EQ bands with 100 mm faders as opposed to the 60 mm ones found on other boards.

Short of recommending the \$80,000 units (money is a slight factor), what is your experience with dedicated boards? Lee Farmer via Internet

You don't need to spend \$80k on a console to have it sound good. A lot of what you're paying for when you get an SSL, AMS Neve, or Euphonix console is the ability to interface with anything, the ruggedness to survive 24/7 operation, and incredible patching flexibility. Plus there's usually head-room for miles. They're a beautiful thing, but most of the time you won't find one for less than the price of a house.

But if you can live with these few sonic and operational limitations (and most of us have to), then small format consoles from manufacturers like Behringer, Mackie, and Yamaha do sound excellent and should serve quite well for your application. There are no inherent "sound signatures" to any of these boards that will make one stand out for drums. But, in the big picture, your sound will be more dependent on mic selection and placement than any console choice. And certainly using a console with long faders and sweepable EQ will make it a lot easier to operate and be more flexible in how you can make it sound. This channel strip equalization is what usually gives consoles

their "personality" and sonic signature. And when you get into the stratosphere of console design, you can debate for days the sound of British EQ, and so on. But what most project studios and live applications need is a decent sweepable EQ section on the channel strip. It's really useful to me when I'm mixing drums since it allows me to really tunein the snare and kick.

As for mic selection on drums (I know you didn't ask, but every engineer has an opinion on everything), I like to use Sennheiser 421's or E604's on racks and floor toms, a Shure SM57 on the snare, a Crown CM 700 on hihat and sometimes overheads, and a Sennheiser 602, Electro-Voice RE20, or AKG D112 on kick. Of course, this changes all the time depending on my mood and what happens to be in the mic kit. Plus the type of music you're mixing has a lot to do with this. For instance, I've loaned out my Sennheiser 602 to the guys mixing Ziggy Marley, and they said it was perfect for reggae kick. On the other hand, it's too bottom heavy for the zydeco acts I've done that really need an SM57 without any subbottom. I may use an RE20 with live sound systems that don't have a huge bottom to begin with, but on some systems it's a little "wooly" off the get-go, so I'll put in a D112 instead for a little less upper bass. The list (and choices) go on and on.

> Mike Sokol EQ Magazine Contributing Editor



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CIRCLE 73 ON FREE INFO CARD

EQ&A QUESTIONS & ANSWERS

STREAM TEAM

I'm a little confused on a few points: [1] If you put your music on a Web site, are you expected to provide MP3 and streaming audio files? [2] Do you need to have two separate applications to create MP3 and streaming audio files? [3] The FezGuys say that, if you plan to make streaming audio files, you need to rip as a WAV file...I was confused as to whether you create the MP3 file from the WAV file or whether that's done totally separately. Will you wind up with streaming files (WAV) and MP3 files on your computer? Guy

via Internet

Different upload sites request you send your music in different ways. The most common these days is the MP3 format, though some may also accept RealAudio. Others may actually request you send them your CD so they can encode

it themselves. Secondly, the average user is going to use two different encoding applications to create RealAudio and MP3 files. If you have some money to spare, you can purchase a software package such as Terran's Media Cleaner Pro (www.terran.com), which easily produces both MP3 and RealAudio, as well as other file formats, like QuickTime. Finally, in the method outlined at www.fezguys.com/features/2000-02/ gigmag/, we're using Xing's Audio-Catalyst (www.xingtech.com/mp3/ audiocatalyst/) to rip to a WAV file, and again using AudioCatalyst to encode the WAV file to MP3. The Real Networks RealEncoder (www.real.com) is used to encode to RealAudio. If you were only creating an MP3 file, you would rip directly to MP3 with AudioCatalyst, bypassing the WAV file process. Good luck! FezGuys

EQ Columnists

THREAD OF THE MONTH

As seen on the EQ Boards at www.eqmag.com

What does everybody use to defrag their hard drives and how often do you do it? Ever do it in the middle of a session? —Dave

If you use Windows 9x, then I'd recommend only using the built-in defrag utility. With NT 4.0, DiskKeeper light should be used. Windows 2000 has DiskKeeper light built-in (how handy). If you use a Mac, then you probably want to check out Norton Utilities.

However, Norton is the devil on PCs, IMHO. It completely takes over and installs components that you specifically tell it *not* to install. I used to use Norton all the time, but each release becomes more and more like Big Brother.

As a general rule, I usually avoid any third-party system utilities (Uninstall, System "Cleaners," Virus Protection) apps like the plague. I keep my audio on a separate hard drive than what my operating system is using. I usually defrag both drives once a day either before or after, but never in the middle of a session mainly because defragging can take a while. —Dylan Walters

In my experience, defrag only helps if there are no edits in your audio files.

If you have perfectly defraged files, but lots of edits and choruses pasted around, the hard disk is going to be seeking its brains out anyway.

The other thing to think about is that the defrag utilities do not know what au dio files go with what session, so you could end up with a worse case than you started with. Each sound file may be defragmented, but the other half of the stereo piano might have been moved to the other side of the disk.

I usually do defragmenting by copying to a different drive. This keeps files together in a folder because they are copied in order to the new drive. —Roger Nichols

Actually, I've heard that file interleaving is much better than defragmenting for hard drive audio files. Go to <u>www. analoguex.com</u> for an interleaving utility and a detailed explanation about why it's better than defragmenting that is way too long to explain here. Plus that utility, and all the rest of his software, is free. —Bob

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"24-bits is an upgrade?" (No, all the bits are there too.)

"Do I have to pay extra for inputs and outputs?" (No. At \$3,999 estimated street price* you get a full set of 24 TDIF-1 or ADAT[®] optical digital inputs and outputs — plus an assignable stereo AES/ EBU - S/PDIF pair For a little more you can get 24 channels of AES/EBU digital I/O, or analog — or both digital and analog!)

"Does it need an external computer?" (No. The MX-2424s front panel has a full set of professional

transport, editing, and track assignment controls, including a shuttle/ scrub knob. So you don't have to have a computer to run it. But — if you happen to own a Mac or a PC, you can take advantage of the digital audio editing and control software that comes standard with each MX-2424 to do even more. Your choice.)

"Before I start recording do I need to buy a monitor, a keyboard, or a hard drive? Or anything else?" (No. Nyet. Nope. Not at all. Just hook up power and start recording.)

So let's make this as plain as we can: The MX-2424 is an amazing, full-featured professional 24-track digital recorder. And there's never been anything like it at this size or price.

ts sonic performance is outstanding. Lots of companies claim 24-bit 48k performance, but only the MX-2424 is part of TASCAM's M Series family of multitracks — the products chosen for their sonic performance by such discriminating facilities as Skywalker Sound, Universal Studios, and 20th Century Fox.



\$3,999^{ESP*}.

S uperior reliability is guaranteed. The MX-2424 was designed from the bottom up to be a great recorder, and nothing but a great recorder. Its processors and circuitry are fully optimized for audio - not video games, spreadsheet software, or surfing the web. And isn't that absolute focus and rock solid performance exactly what your music deserves? Over the last three decades we've designed and built literally millions of professional recorders and recording systems; the MX-2424 is the culmination of everything we've learned.

So easy to operate, you could do it blindfolded. Of course that way you'd miss the great light show from the 24 tracks of level metering and channel status displays... but the real point here is simplicity. When you want the MX 2424 to start recording, just reach over and press REC + PLAY (just like a traditional tape recorder). In a fast-paced production environment, you can record to hard drives that mount into standard Kingston[®] carriers and plug into the front panel drive bay. Just pop in a new drive at the start of each session. It doesn't get any simpler than that.

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Really.

The power to meet your needs. A standalone MX-2424 is an incredibly powerful unit, with enough internal hard disk capacity to hold about 45 minutes of 24-bit 24-track audio. The MX-2424's Fast/Wide SCSI port lets you connect up to 15 external drives and record directly to all of them. And if you need more than 24 simultaneous tracks, just add additional MX-2424's. Up to 32 MX-2424's can be locked together in sample accurate sync to act as a single recorder.

Professional recorders need to interface with increasingly complex systems.

✓ It provides video and time code lock capabilities as standard features, making it easy to integrate with external workstations.

✓ It resolves to AES/EBU, S/PDIF, word clock, TDIF-1, ADAT optical, SMPTE Time Code (LTC), and video, and chases MIDI Time Code.

✓ Available Input/Output modules include TDIF-1, AES/EBU, ADAT optical, and analog. It's a complete professional hard disk multitrack in a portable, affordable, rackmount box. You can plug it in, turn it on, and start recording.

✓ Back panel ports include Fast/Wide SCSI, ethernet, MIDI, RC-2424 remote, and TL-BUS!

xtend your reach --Want a remote control? Get the one that's made to take advantage of the power in your MX-2424. The RC-2424 is a powerful, remote professional multi-machine controller with all of the MX-2424's front panel feaplus macros and tures, more.

MX-2424 shipments are about to start, and there is already a waiting list. To get yours sooner instead of later, contact your authorized TASCAM dealer!

*So... what's this Estimated Street Price? Instead of quoting you some meaningless "List Price," ESP is what we expect typical U.S. customers to actually pay for an item. It gives you a better way to compare value when you shop.



A Whole World of Recording

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CIRCLE 58 ON FREE INFO CARD World Radio History

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Music and Internet Expo Takes New York by Storm

The big music-related event in the Big Apple during early March was the second annual New York Music and Internet Expo. Headlined by a raft of star keynoters (including Robert Fripp, Nile Rodgers, Thomas Dolby Robertson, and rapper/actor Ice T), the three-day event drew a crowd of over 5,000 attendees. Veteran music executive Danny Goldberg (former head of Mercury Records) made the biggest splash, unveiling a new Internet venture called ArtistEnt. In a sea of new Web offerings, this one is notable because it has purchased Todd Rundgren's PatroNet an online music distribution and

marketing service that also supports the creation of subscription-based Web sites that, in theory, should pay for themselves. What's more, ArtistEnt will provide an online presence for artists on the roster of Goldberg's indy label, Artemis Records, as well as allowing other artists to upload new content weekly (already signed up are Sugar Ray, Peter Wolf, and Rosanna Arquette, with more to come).

No less than 15 panels were held during the three-day event. The panelists represented a broad cross-section of the New York music industry both mainstream and alternative —





including legen dary Pink Floyd/Alice Cooper producer Bob Ezrin, rapper Chuck D, ex-Talking Heads Jerry Harrison, Chris Frantz, and Tina Weymouth, and ex-Blackheart Ricky Byrd. These provided the framework for fast and furious discussions on

2000

Steinberg Launches the Cubase Network

Steinberg, a leading provider of professional music and audio software, recently launched the Cubase Network with the first publicly available Internet Recording Studios at <u>www.</u> <u>cubase.net/studios</u>. Musicians and audio professionals can meet online to collaborate and produce original audio from anywhere in the world.

Users of Steinberg's flagship product, the digital audio sequencer Cubase VST, can now download Cubase with RocketPower for no charge. This allows for multiple Cubase users to work together and share projects using the Internet as a live connection - all in the familiar Cubase VST environment. Entry to public studios is free, and a limited number of private studios are also available for free (the first 100 Cubase users who sign up). Additional private studios are available for \$29 (one month) and \$99 (three months).

All members of a session can work in real-time together, hearing the same work-in-progress as it develops. Musicians can post one of their own projects and ask for creative contributions, or they can join a session and contribute to someone else's music. The Internet Recording Studio is used to update each project as it develops in real time. All participants work on their own aspects of the same arrangement with complete control over how and when their contributions will be heard, because they can publish their changes whenever they like.

For more information, check out www.cubase.net/studios.

hot topics such as copyright protection, database marketing, "grassroots" promotion, and the future of Internet commerce, including radio and TV Webcasting. All in all, the event provided vital information for the musician of the 21st century, who may well find him or herself wearing even more hats than the simple singer/songwriter/producer/ engineer of the late 20th century.

According to the promoters (perhaps rather optimistically), next year's Expo is planned to be held in Madison Square Garden. Stay tuned right here for details as they become available. —Howard Massey

World Radio History

APRIL

The Revolutionary Way to Mix, Master and Make CDs



9600

The new MasterLink" ML-9600 high-resolution recorder is much more than just a mixdown deck or CD burner. It's a visionary product that will completely change your perspective on two-track audio, and redefine the way you master your mixes.

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Create 16 different playlists containing up to 99 songs each ...with full control of song order, track gain, fades and more.





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Record your master to CD-R. MasterLink creates standard 16-bit/44.1kHz Red Book CDs as well as AIFF-compatible hi-res discs up to 24-bit/96kHz.



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Fine-tune each song with

MasterLink's 3-band, fully

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QSC Expands Online Technical Library

Contractors can now download drawings of the latest QSC products from QSC's newly expanded online Technical Services Library. The drawings, which are in the DXF format, can be imported into AutoCAD and other computeraided design programs to assist contractors in creating front and rear rack elevations. Files can be accessed directly by visiting www.qscaudio.com/ support/library/cad/cad.htm.

Customers can contact QSC's Technical Services Department at 800-772-2838 or they can visit www. gscaudio.com.



Grotto Studios Install Panasonic DA7's in Tandem

Los Angeles-based Grotto Studios has installed and linked two Panasonic DA7 digital mixers together, giving the ability to mix up to 76 channels of digital and analog audio simultaneously. Working seamlessly with the studio's multiple ADAT setup, the DA7's are the center of a 48-track fully automated digital and analog recording environment (32 channels of digital audio and 16 channels of analog).

With recently completed projects by Bonnie Pointer (of the Pointer Sisters), the Riverdance troupe, and the Welsh Choir of Los Angeles, the newly remodeled studio houses a 600-squarefoot main recording room with oak floors, sloped ceilings, an isolation booth, and a nine-foot Steinway grand. In the control room itself, the DA7's small footprint stands in contrast to its huge feature set, giving artists, engineers, and producers room to work comfortably and efficiently.

Grotto owner David Williams found that the linked DA7's worked together flawlessly from the start, with increased mixing functionality and automation capabilities over previous analog consoles. These features enable Grotto to offer clients a recording environment that allows more focus on the music, the artist, and the project itself.

John Lennon Contest Winner Records at Ahhsum Lawson

Well over 1,000 contestants entered the John Lennon Talent Contest in Germany. When it was over, Leisure Suit Nerds (LSN) were crowned as the winners. One of the prizes was an all-expense-paid trip to California and full access to Ahhsum Lawson Studios, where they recorded three original songs. Ricky Lawson, Grammy Award-winning composer, producer, and drummer was a gracious host and enjoyed working with the band.

LSN has been together eight years. Rik Larson, the band's manager, says, "We were excited to work here because of the level of professionalism. We knew it would be great, and the weather isn't bad either."



"I was drawn to the DA7 by its flexibility, automation, scene memory capabilities, affordability, and 24-bit sampling rate," says Williams. "The sound quality of the console is just amazing."

For more information on the DA7 console, visit <u>www.panasonic.com/</u> proaudio.

Kind of Loud Technologies Receives Funding

Kind of Loud Technologies recently announced that it has accepted funding from Chance Investments, a privately held equity and investment company with key holdings in the Internet and entertainment industries.

Richard Wolpert, founder of Chance Investments, is currently CEO of CheckOut.com and former president of Disney Online. Chance Investments has also taken an equity stake in Universal Audio, which manufactures authentic reproductions of classic analog recording equipment.

"Surround sound is clearly a key component in future media technologies," says Wolpert. "Kind of Loud is well poised to lead advanced technological development in this area."

For more details, visit <u>www.</u> kindofloud.com.

DIGITAL CONTRACTOR OF THE CONT

Full input module with a knob for every function



96 kHz 24-bit capable

Introducing the Sony DMX-R100: a smallformat digital mixer inspired by our Oxford console, considered by many industry leaders as the most advanced digital mixing system ever developed.

How does the DMX-R100 work? The way you want it to. You have a full input module with a knob for every function. Equalization and dynamics can be adjusted simultaneously. Your hand goes intuitively to the right knob. Your mixing session goes faster. You can concentrate on the mix, not on the technology.

The R100 can memorize your automation moves the moment you touch the highresolution touch-screen fader. Don't tell the mixer to change modes. Don't think about it at all. Just touch it.

A color touch-screen is built into the control surface. Use the built-in router to assign inputs to faders. Select buses, sends, and directs to analog and digital outputs. View a complete input module or zoom in on the EQ and Dynamics sections.

Machine control with 9-pin and MMC interface is standard.

Right out of the box, the R100 is smart enough to make you more productive. And open up opportunities for working in new high-resolution formats, without expensive upgrades or difficult learning curves. Which makes it an educated choice for audio professionals everywhere.

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THE CFX/SRM450 COMBO Ultimate performance & convenience.

t's easy to transport, simple to set up and it delivers a level of accuracy that's never before been possible with a compact sound system. Just plug a CFX•12, CFX•16 or CFX•20 mic/line mixer into a pair of active SRM450s, the first sound reinforcment loudspeaker accurate enough to sound like a studio monitor. Only way WAY louder.

The first effects mixers you won't be embarassed to be heard in public with. Our CFX Series™ start with a typically feature-laden low-noise high-headroom Mackie mixer design...and then add ultrarealistic EMAC[™] 32-bit digital effects, derived from our Digital 8•Bus recording console. Plus a lavishly over-engineered 9-band stereo graphic EQ that doesn't degrade sound quality the way cheap ones do.

CFX Series mixers are easy for non-technoids to set up and use, yet are packed with features seasoned pros appreciate, such as 3-band EQ with sweepable midrange, variable effects parameters, and a way-cool Break Switch that mutes all channels and automatically switches to tape input (which has its own level control, no less).

> Active accuracy: better than powered; way better than passive. Our SRM450 active speakers are re-defining "small" PA sound performance. First and foremost, they're incredibly accurate

SRM450 ACTIVE 2-WAY SR LOUDSPEAKERS:

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- Built-in electronic crossover + time and phase correction circuitry
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- 12-inch LF transducer outplays most passive 18-inch systems
- Built-in mic preamp with level setting control & LED + line input
- · Contour EQ and remote turn-on feature
- Rugged trapezoidal cabinet with three balanced handles for easy transport

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The highest fidelity compact SR system available today. Just add mics, instruments, stands and a modicum of talent.



with crisp, clean treble, sweet natural midrange and loads of tight, low bass. Second, they're capable of awesome output without a hint of distortion. And finally, SRM450s have ultra-wide, even dispersion at all frequencies...so everyone hears the same great sound.

The reason? Custom-transducers that we make ourselves, a unique multi-cell aperture horn design, true active servo-coupling between transducers and internal FR Series[™] highcurrent amps...and electronic time and phase correction.

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If you're a good musician, you'll sound better. If you're an OK musician, at least those wrong notes will be *really* loud and *really* clear.



Made by Mackods in Woodnynie, USA and Reggio Emilia, Italy.



- True 4-bus design
- I6 built-in 32-bit EMAC[™] effects, each with two variable parameters
- 9-band stereo graphic EQ
- 3-band EQ with swept mid on channels with mic inputs
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- 60mm log-taper faders
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- Extra utility output with level control
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- Level-setting LEDs on all mic channels
- 18dB/oct low-cut filters on all mic channels
- BNC lamp socket & EFX footswitch jack

CFX • 12 12 total channels • 4 buses • 8 mic/line channels • 2 stereo line-level channels • 8 channel inserts

CFX • 16 16 total channels • 4 buses • 12 mic/line channels • 2 stereo line-level channels • 12 channel inserts

CFX • 2D 20 total channels • 4 buses • 16 mic/line channels • 2 stereo line-level channels • 16 channel inserts



CIRCLE 31 ON FREE INFO CARD



ADDICTED

udio-Technica announces the introduction of the Sound Addict line of microphones, consisting of the Bark vocal mic and the Bite instrument mic. Both models feature a dvnamic element, a cardioid polar pattern, a frequency response specifically tailored for their particular application, a low-impedance output, and an integral XLR connector. The Sound Addict mics have an industrial look with grooved metal handles and stainless steel color accents on the matte black finish. Each Sound Addict microphone comes in a futuristiclooking, molded black plastic case with a screw-off end cap. For more information, call Audio-Technica at 330-688-3752 or visit www.soundaddict.com. Circle EQ free lit. #116.





LIVE POWER

SC announces the new PowerLight 2 Series, the latest amplifier line exclusively designed for touring and live sound professionals. The line's four initial models the PL 218, PL 224, PL 230, and PL 236 — range in power from 900 watts to 1,850 watts per channel (2 ohms) in a 21 lb., 2U chassis. Built-in clip limiters and selectable lowfrequency roll-off filters boost system response and ensure low-end tightness. A standard Data Port allows users to interface with QSControl, QSC's

computer control system, or other audio network control devices. Other features make the amps ideally suited for touring use in smoky or dusty environments. All models include a detachable "locking" IEC power cable and offer a variety of I/O options, including Neutrik Speakon and binding post outputs and heavy duty Canon XLR and 3-pin detachable Euro-style input connectors. For more information, call QSC at 800-854-4079 or visit www.qscaudio.com. Circle EQ free lit. #117.

MULTIPURPOSE MIC

he pro audio division of Marshall Electronics is introducing the MXL600, a newly developed 20 mm gold-diaphragm condenser microphone in a miniature body. Designed for professional recording applications, the new mic has the dual qualities of a large-diaphragm condenser and a small-instrument mic. A newly developed pre-aged 3-micron diaphragm with a 20 mm diameter is formed into a 22 mm screw-on capsule that fits directly on top of a 90 mm long x 22 mm body, which contains a FET preamp and transformerless output circuit. The product also boasts a 40 Hz–20 kHz frequency range, 47 dB S/N ratio, 20 dB equivalent noise level, and 134 dB SPL level. It runs off of 48-volt phantom power. The MXL 600 is available for \$269.95. For more information, call Marshall Electronics at 800-800-6608 or visit www.mars-cam.com. Circle EQ free lit. #118.



MASSIVE PASSIVE STEREO TUBE EQ



A PICTURE IS WORTH A THOUSAND WORDS...

Perhaps, but would photographs of our Variable Mu or VOXBOX have created their successes alone?

You have to hear this gear. You have to use this gear. Put your hands on the knobs and crank 'em.

MANLE

Engineers who have already gotten hold of the MASSIVE PASSIVE have told us: "Why does it make everything sound so much better?", "It's organic and orgasmic.", "It's a f%#king powerhouse.", "It's unlike <u>any</u> other EQ.", "This is IT. The sound I've always dreamt of but couldn't ever get until now."

GOT THE PICTURE?



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Craig 'HUTCH' Hutchison designed these monsters... The MASSIVE PASSIVE is a two channel, four band equalizer, with additional high pass and low pass filters. "Passive" refers to the tone shaping part of this clever new EQ design not using any active circuitry. Only metal film resistors, film capacitors and hand-wound inductors sculpt the sound, kinda like a Pultec EQ on hyper-steroids. Super-beefy, hugelyhigh-headroom Manley all-tube make-up gain amplifiers deliver your tunes into the next realm. You'll need to experience this.

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KING OF POWER

rown is launching the new CE 4000, its highest-powered CE Series amplifier to date. Utilizing a switching amplifier design rather than a linear amplifier design, the CE 4000's BCA topology delivers power while generating just one-tenth the heat of a conventional amplifier. The CE 4000 delivers a solid 1,800 watts per channel (both channels driven) into 2 ohms, 1,200 watts into 4 ohms, and 600 watts into 8 ohms. The CE 4000 features front-panel level controls, useful function indicators, proportional fan-assisted cooling, short circuit protection, and a rugged 3U metal chassis. For more information, call Crown at 800-342-6939 or visit www.crownaudio.com. Circle EO free lit. #119.

SIMPLE SPEAKER

acPherson, Inc. introduces the AXIA loudspeaker system, which is designed for fixed installations or touring. Available as a triamp or three-way biamp system, the AXIA's design allows cabinets to be used alone or coupled together to form a full-range vertical line array. The AXIA premium Baltic birch houses one 15-inch 1 F, two 8-inch MF, and a 2-inch HF mounted on a 90- x 55-degree slot-CD horn flare. AXIA's ExoSpine Rigging System integrates the flying hardware directly into the cabinet, making setup and tear down simple. For more information, call MacPherson at 847-674-3535. Circle EQ free lit. #120.

BETA TEST

hure's Beta 87C microphone is available both as a wired microphone and as a capsule for use with Shure wireless systems. The Beta enables users to effectively reject ambient noise arriving at the rear of the microphone capsule. Other features of the high-output Beta 87C include high gainbefore-feedback and a low-frequency roll-off that provides greater command of proximity effect. The microphone offers a frequency response of 50 to 20,000 Hz, a dynamic range of 117 dB, low distortion characteristics, minimal off-axis tone coloration, and the ability to handle sound pressure levels up to 139 dB. Supplied accessories include a built-in OPC filter, swivel adapter, and carrying/storage bag. For more information, contact Shure at 847-866-2200 or visit www.shure.com. Circle EO free lit. #121.

IN THE MIX

tudiomaster has launched the new Club 2000 Hi Z Plus series mixers, replacing the popular Club 2000 and Club DSP range. The Hi Z Plus retains all of the professional features of the Club 2000 range (such as the 3-band mid-sweep EQ), while

providing an upgraded appearance package and a newly redesigned input stage. Dubbed the "Hi Z Plus," the mixer's mono and stereo line inputs now boast an expanded operating range and will effectively accommodate a variety of different high-impedance sources. The mixers range in price from \$449.95 to \$679.95. For more information, call Studiomaster at 714-998-2102 or ***** visit www.studiomaster.com Circle EQ free lit. #122.

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The Pro Suite provides an arsenal of software tools for today's recording musician. It combines essential recording and sampling software technologies into an integrated studio solution. There's nothing else like it available today.



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Notation with guitar tablature, fretboard editing
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CAKEWALK AUDIO FX " 1 DYNAMICS PROCESSING

- Compressor/Gate maintains audio signal levels at user-defined levels
- Expander/Gate increases dynamic range of audio
- Limiter prevents audio signals from exceeding user-defined threshold

CAKEWALK AUDIO FX" 2 VINTAGE TAPE AND AMP SIMULATION

- Advanced processing algorithms add classic sound and warmth to "dry" and "cold" digital audio tracks
- AmpSim adds guitar amplifier sound to digital audio; choose and modify amp model, speaker cabinet, overdrive, EQ and other parameters
- TapeSim adds tape saturation and natural warmth associated with analog magnetic tape decks





CAKEWALK AUDIO FX" 3 SOUNDSTAGE DESIGN FOR CUSTOM REVERB

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All Cakewalk Audio FX plug-ins provide:

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- SoundFont instruments
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- More

Cakewalk Pro Suite is available worldwide. For more information, visit www.cakewalk.com, or call 888-CAKEWALK (617-441-7870 outside U.S.).





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cakewalk

Rocket Network

The recording studio networking service arrives with the potential to change the way we collaborate on projects

BY STEVE LA CERRA

Initially announced several years ago, Rocket Network is officially online and operating, providing audio networking services to recording studios, engineers, and producers via the Internet. Rocket Network is the first (and currently the only) company to offer "Internet Record-

ing Studios," whereby audio professionals may meet in online studios for collaboration on, and storage of, recording projects. By offering an alternative to the traditional recording environment, Rocket Network can help reduce production costs as well as expedite completion of recording projects.

Central to the Rocket Network concept is the Internet Recording Studio, a virtual work-

place where engineers, producers, and composers in varying locales can exchange ideas while working on any audio and/or MIDI-based production. Rocket Network leases clusters of Internet Recording Studios to third parties such as audio software manufacturers. Their customers can then access Rocket Network Internet Recording Studios via the software they are already accustomed to, in a partner's own-branded Studio Center. For example, Steinberg Cubase users can access the Steinberg Studio Center of Rocket Network Internet Recording Studios through www.cubase.net.

Similarly, Logic users can access the Rocket Network via Emagic's Studio Center, which is hosted by Harmony Central. Each Studio Center contains both public and private studios. Entry to "public" studios is free of charge for registered users; those who like the concept, wish to store their projects, and want to restrict access to the files can purchase "private" studio space through the partner-company. Entry to private studios requires a modest fee annually (upwards of \$99), depending on bandwidth requirements.

ROCKET CONTROL 2.0

Access to the Rocket Network is facilitated through an application called RocketControl 2.0. Available for both Mac and PC, RocketControl 2.0 is a free download (available at <u>www.rocketnetwork.com</u>) that provides communication to, and navigation of, the Rocket Network. It also acts as a soft-



ware "bridge" between the Network and any RocketPower application a user might be running. **RocketControl 2.0** provides public and private chat and navigation hot links, and includes support for a variety of audio file compression formats such as MP3 and Odesign Music CODEC. The RocketControl

Professional Upgrade allows users to access private studios, bookmark users and studios, and edit their online user profile for a searchable talent database. As of this writing, Emagic's Logic Audio and Steinberg's Cubase VST were available with RocketPower; Digidesign, Euphonix, and Gvox have announced plans to create RocketPower versions of their products.

HOW IT WORKS

Registered users of the Rocket Network log onto a Studio Center using a unique user name and password. Upon verification, the RocketControl Web Interface allows the user to browse through the available Internet Recording Studios within a Studio Center and choose one at which to work. Basic information about each Internet Recording Studio such as name, studio owner, and access privileges available are displayed on screen. If the studio is currently empty, the user may "post" (upload) a new project to share with other audio pros. Each post automatically updates the master arrangement and distributes the new parts to people working on the project. Whenever a user enters a studio where there is a project in progress, the most up-to-date arrangement is offered for download. Once downloaded, the user can create and record a new track for the project using their audio application. After the track has been finished, continued on page 126

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JOEMEEK

If you haven't tried the new Joemeek Channels, you haven't got a Q!

Joemeek set the standards for "vintage" retro recording designs by offering quality and value. No other direct recording channel can match the silky smooth colored sound of Joemeek. While other units claim to reproduce colored "vintage" sound, wouldn't you rather own the one that does.

got meek?

CIRCLE 23 ON FREE INFO CARD

KRK Di Series Monitors

KRK offers a glimpse at the future of monitors with the FireWireequipped Di Series

BY STEVE LA CERRA

One of the unique products introduced at the Los Angeles Winter NAMM convention a few weeks ago was KRK's new Di Series of studio monitors. Di Series loudspeakers are the first monitors to employ FireWire ports (also known as IEEE1394 or i.LINK to you tweak heads) for digital interface to the external audio world. In the Di Series monitors, audio is processed "on-board" using proprietary DSP to perform digital equalization, crossover, and time-alignment functions. Building upon their well-established line of active loudspeakers, Di Series technology will initially be incorporated in KRK's Exposé E7 and E8 monitors, ultimately branching out to the entire KRK line. KRK is also offering current owners of the Exposé E7 and E8 the opportunity to have their monitors upgraded to Di technology.

In addition to the FireWire connection, the E8Di and E7Di feature a hardware and software digital audio solution developed by SoftAcoustik of Quebec. Taking advantage of the fact that audio is streamed to the loudspeaker while still in the digital domain, the Di Series employs DSP-based crossover and equalization functions prior to the digital-to-analog conversion process. For this purpose, SoftAcoustik has developed the SA 1.0 digital engine, a programmable, ultra-high-quality audio-processing algorithm that claims to provide extremely accurate frequencyand time-domain response. By performing crossover functions digitally, the phase errors typically created in the crossover process can be eliminated, and speaker equalization can be fine-tuned while maintaining noise levels lower than those associated with analog equalization.

Additionally, precise time-alignment of the drivers can be performed using the DSP. Benefits obtained from using the SA 1.0 digital engine include improved overall accuracy and dynamic response, a deeper sound stage, and a lower noise floor. Post-DSP audio is converted for amplification via 24bit/96 kHz digital-to-analog converters. For audio sources producing Lightpipe, S/PDIF, AES/EBU digital, or analog audio signals, a "gateway" interface box will be introduced that handles the conversion to FireWire protocol for input to the speaker.

The analog features of the biamped E8Di include an 8-inch Kevlar woofer and a 1-inch inverted-dome Kevlar tweeter. For those unfamiliar with the material, Kevlar is an extremely rigid, lightweight material that is virtually indestructible (one of its other applications is in bulletproof vests). In addition to being stable under varying tempera-

ture, humidity, and UV conditions, Kevlar's resonant frequencies lie outside the audio bandwidth. allowing a smooth frequency response. Each amp in the E8Di provides 140 watts per driver, enabling the monitor to achieve an average SPL of 114 dB and a peak SPL of 123 dB. The E7Di specs out similarly for power handling and SPL, though it employs a slightly smaller 7-inch woofer for a 3 dB down point of 54 Hz (as opposed to the E8Di's 3 dB down point of 45 Hz). In both the E7Di and the E8Di, drivers are individually protected against overload conditions.

By dealing with audio in the digital domain, the E7Di and E8Di are capable of performing functions not associated with traditional "analog" monitors. One example is self-analysis and tuning of the speaker. Any speaker manufacturing process results in minute variations of frequency and phase response from cabinet to cabinet. Through use of a calibration microphone and audio analysis program, characteristics of a Di speaker can be examined and compared to a standardized reference. Data collected from such analysis would then be downloaded to flash memory employed in the speaker's DSP, enabling KRK to tolerance-match each speaker.

In 5.1 listening rooms (pro or consumer), a single FireWire loop may be

continued on page 122





HOM WICH WIXER DO YOU NEED?



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CIRCLE 63 ON FREE INFO CARD

Men of DaHouse

DaHouse Productions provides a variety of audio services under one comfortable roof

STUDIO NAME: DaHouse Productions **LOCATION:** Ringgold, GA

KEY CREW: Kent Silvey, multi-instrumentalist, singer/songwriter, owner, and president of DaHouse Productions; Jo Whitaker, drummer, percussionist, chief engineer, and co-owner of DaHouse Productions; Sam Bolden, guitarist, songwriter, and co-owner/CEO of newdayrecords.com. Silvey, Whitaker, and Bolden have a musical history of more than 15 years together; their forthcoming CD will be available at <u>www.</u> newdayrecords.com.

CREDITS: Silvey has recently focused his energies on writing songs and movie soundtracks for major record labels. In addition to producing his soon-to-bereleased third CD project, his music has also been featured on various video networks. Other DaHouse projects include Shaking Ray Levis Society, Adrenalin, Groovy Grapes, Kathy Tugman, Road Kings, Unsatisfied, Gathering, Cabrini Green, Mystery Cycle, Vince Stallings, Robert Crabtree, Eddie Gwaltney, and a variety of commercial jingles, radio, TV, and video spots.

MIXING CONSOLE: Mackie 24x8x2

MONITORING EQUIPMENT: Tannoy PBM 6.5; Alesis Monitor One; Tannoy subwoofer; Hafler Pro 2400 amplifier

RECORDERS: Alesis ADAT XT20 [3] and ADAT; TASCAM TSR-8

DAT MACHINES: Panasonic SV-3700 OUTBOARD GEAR: Roland DEP5 [2] and SDE1000; Alesis Quadraverb, 3630 compressor [2], and Nanoverb; Yamaha SPX90 [2]; DigiTech S-100 reverb; Audio Logic MT44 quad gate; Behringer Autocom compressor [2]; BBE 422A Sonic Maximizer; ART Tube MP mic preamps [2]; Peavey Ultraverb

KEYBOARDS: Roland XP-50 and D-20; Yamaha TX81Z; Baldwin upright piano **COMPUTERS AND SOFTWARE:** Dell Dimension L500R P3/500 MHz with 128 MB RAM, 10 GB drive; Accell Microsystem 400 MHz/P2 with 64 MB RAM and 8 GB drive; Sonic Foundry CD Architect XP4.0 and Sound Forge 4.0; Syntrillium Software Cool Edit Pro SE; Lexicon Core 2 audio recording hardware

MICROPHONES: AKG C414; Neumann KM

86; CAD E-300; Shure SM57 and SM58; Electro-Voice EV408 and EV308

EQUIPMENT NOTES: Silvey states: We have a variety of guitars and basses available here at the studio, including Guild, Gibson, Fender, Ovation, and Yamaha instruments, but we (as well as many musicians who record here) have found that we get cleaner and richer sounds using Warrior instruments. STUDIO NOTES: Continues Silvey: One thing that makes DaHouse Productions so special is the atmosphere and environment. DaHouse is (as its name implies) a house converted into a multitrack recording and production facility offering a creative yet comfortable approach to recording. Many of our out-of-town guests also stay at DaHouse while recording their project here. This is what I call a full-service studio! In addition, we offer a unique service unavailable to most artists, which is the ability to

market, sell, and distribute CDs and merchandise across the world.

PRODUCTION NOTES: Whitaker says: I like to record as live as possible, at times even setting up a wedge for the lead singer to hear. Our studio is large enough to set bands up



the way they are used to rehearsing or performing. It is all about the atmosphere and how comfortable an artist is during the performance in order to capture the right feel. Throw up some good microphones, get levels, and watch us work!





Altec 670B Versatility was this mic's middle name

MICROPHONE NAME: Altec 670B FROM THE COLLECTION OF: William Meredith, Cinesound Company, NYC YEAR OF MANUFACTURE: circa late 1950s TYPE OF MIC: Ribbon

POLAR PATTERN: Cardioid, omnidirectional, and bidirectional (see notes)

FREQUENCY RANGE: 30 Hz to 16,000 Hz (frequency response is stated as 30 Hz to 10,000 Hz ±4 dB)

OUTPUT LEVEL: -56 dBm/10 dynes per square centimeter

OUTPUT IMPEDANCE: 30/50 or 150/250 ohms

HUM: -130 dB referenced to 10⁻³ Gauss **FRONT-TO-REAR REJECTION:** 15 dB average from 30 Hz to 10,000 Hz

DIMENSIONS: 6.5 high x 2.5 wide x 3.375 deep (inches)

WEIGHT: 1.25 pounds

MIC NOTES: Designed for recording studio, broadcast, and public address applications, the Altec 670B uses an aluminum alloy ribbon element coupled to an acoustical labyrinth chamber with a length of 36 inches(!). This acoustic vent permits operation of the mic in omni, cardioid, or bidirectional patterns, as well as intermediate variations (details below). The 670B was introduced as a smaller brother to the Altec 670 in order to meet the increased demand for a lightweight, less-obtrusive microphone.

USER TIPS: A screwdriver-operated switch on the 670B selects between the three "standard" pickup patterns. In addition to this switch, the 670B features a double-slotted "shutter" that can vary the polar pattern to a multitude of variations in between the main three patterns. This makes it possible to shift the mic's null point over a 90-degree angle, and thus "tune out" interfering noises from the surroundings. Output impedance of the 670B is easily changed by removing the Altec nameplate and accessing an internal switch.

Technical data furnished through the courtesy of Bob Paquette of the Microphone Museum in Milwaukee, WI.




CIRCLE 47 ON FREE INFO CARD World Radio History

Anatomy of a Remix

A look at why songs are remixed and how it is done



BY CRAIG ANDERTON

It's one thing to create a song and another to mix it. But it's a whole different experience to have it remixed by others. Two of my tunes were recently remixed for subsequent CD release, and I was fortunate enough to be present during the remix process. For those who haven't been involved with remixing, you might enjoy an insight into how this works...here goes.

THE B.W.A. REMIX

The project involved Cologne's premiere remixers, Dr. Walker (Ingmar Koch, who has done over 50 remixes for various artists and labels) and Thee Joker (profiled in the April 1999 issue on Performing Engineers). As collaborators, they go under the name B.W.A. ("Beerdrinkerz With Attitude"). Walker is a wizard with mixing consoles and beats --- check out his article "Rockin' the Mixing Board" in the February 2000 issue of EQ — while Joker comes from a bassist/DJ background. This was a promising start: with Walker specializing in the mixing decisions and Joker being more concerned with pacing and

flow, it showed once again that getting a good DJ involved with your music can give it an extra edge and perspective.

The two tunes were based on samples from my upcoming CD (Sexy World, which will also be available as a loop library for Sonic Foundry's Acid program), and were remixed in Acid. With the loop library, the loops are arranged in a folder that contains all samples used in a given tune, the Acid file that arranges them, and a mixed audio file. The process started with Walker listening to the mixed file to get an idea of the available samples. Although the first tune to be remixed (called "Dogs") initially had a tempo of 130 BPM, he slowed it down to 120, giving it more of a dance/funk feel than the more techno-ish original version. The other tune, "ZepHop/NYC - 01" was sped up from about 100 to 120 BPM.

He then previewed the individual loops, and dragged anything he liked from Acid's Explorer-style interface into the main arranging window. However, before setting up the rhythm tracks, Walker chose only loops involving melody lines or melodic phrases, and worked on each lead. I had expected the emphasis to be solely on the rhythm, with melodies as almost an afterthought. But, as with conventional songs, the lead lines received the most attention in developing the sound.

To provide the required cluboriented rhythm track, Walker initially dipped into his personal library of beat loops. My original rhythm track

loops were used only toward the end, when the tune needed an extra push (although several "lighter" tracks containing individual percussion parts were brought into the mix early on).

One of Walker's favorite tricks is to use Acid's ability to time compress or expand the loop's *initial* length (of course, Acid can also do this "on the fly," but here the application is somewhat different). For example, one key loop in one of the tunes was built on a phrase from the Ensoniq ASR-X that contains the word "dog." The original loop was four beats long, but, by stretching it over eight beats, the "dog" got stretched out big-time, making a comical effect even more comical. That was the point where the remix concept was established: focus on the "dog" phrase, and have it serve as the musical equivalent of a punch line. Later on, some other phrases were also stretched from four to eight beats, providing more "drawnout" sounds that worked well with the main phrase.

I GOT RHYTHM

After selecting and tweaking the leads, Walker extended the loops for about half a minute. As the point of the remix was to turn the tunes into slammin' dance floor tracks, a crucial part of the process was selecting the all-important rhythm track loops. The looped leads provided a reference against which he could preview the rhythmic loops. Acid's preview function is particularly useful; you can audition the loops at the correct tempo, in context with the other tracks.

Walker went back and forth with Joker on which new loops worked well with the track and which didn't. The ones that made the cut were again dragged into the main window. The "primary" loops that formed the backbone of the rhythm track were extended, providing a backdrop that made it easy to tell if the supporting loops would work well with them.



FIGURE 1: The area where the sixteenth-note loop "intros" were added appears in inverse color. Note how the sixteenth-note loop fragments lead into the main loops.



four views of Creation

Ricky Martin's *Livln' La Vida Loca* was the first No. 1 single recorded and mixed entirely on Pro Tools — that says it all right there. We did everything in Pro Tools including editing loops, adapting textures, fusing takes, and using AutoTune and VocALign for the vocals. Every single note was done on Pro Tools. With Pro Tools, we can do things that are impossible to do using tape.

Desmond Child Producer, Ricky Martin | Cher | Hanson

Pro Tools, SampleCell TDM, and TDM Plug-Ins have long been the core of my creative platform, and with the addition of the Virus TDM Plug-In and Koblo synths with DirectConnect, I've got unbelievable synthesis power that occupies zero rack spaces. Pro Tools, SampleCell TDM, and TDM Plug-Ins were an important creative package during the making of *The Fragile*.

Charlie Clouser Keyboardist, Nine Inch Nails

These artists are changing the way music is made by lurning the studio into a creative instrument. Of course, they're doing it with Pro Tools — the ultimate audio production system. Pro Tools gives you creative possibilities that far surpass any tape-based console, with DSP muscle no sequencer application can touch. With options like Koblo Studio9000, SampleCell II Plus,[™] Virus TDM Plug-In and DirectConnect, Pro Tools gives you everything you need to take your music from idea to finished masterpiece.

CIRCLE 13 ON FREE INFO CARD

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Using Pro lools has allowed us to be much more creative and flexible in the studio. You can kind of forget about the recording process and, ultimately, be way more creative. I can't imagine going back and being forced to record everything with the limitations of analog tape and the old mixing console. Once you've used Pro Tools, there's no turning back.

Butch Vig Producer, Gendage | Smashing Pumpking | Nirvana

> Pro Tools creates a whole new world where I can see my music. It's infinite as far as what I can do, especially for hip-hop music. I can chop in drum breaks and different beats so easy using Pro Tools. And the new version 5.0 MIDI sequencing features are awesome. With Pro Tools I can do anything I imagine.

> > DJ Lethal Turntables and semplify

WHY SHOULD/COULD/WOULD SOMEBODY REMIX **AN EXISTING TRACK?**

We asked Dr. Walker the preceding question, and he gave us these answers:

- We like the track and want to make it better.
- We don't like the track and want to make it better.
- We like the track and we want to add our own style and ideas.
- We like the artist and we want to do him a favor.
- We don't like the track and we want to make it worse.
- We don't like the artist and we want to kick his ass in a musical way.
- let's do a remix."
- We want to make money.
- We have to make money.
- We thought it would be fun to do after having way too many beers.
- Or, finally, a mild or wild mixture among all these options.

ARRANGING

After selecting the various loops, it was time to arrange. Although Acid makes it easy to arrange loops on the fly, this was still the most tedious part of the process, as the tune's flow is crucial. Therefore, many times after making any kind of significant changes, Walker and Joker

replayed the song from the top to make sure that the flow was maintained throughout. Unlike the deliberate pace of mixing most pop or rock music, loops were quickly selected, discarded, extended, or shortened without second thoughts. We were all completely in right-brain land, with no intellectualizing

and hardly any discussion: if something worked, it was instantly obvious and, if not, that was equally obvious, usually within seconds.

This is where Joker's expertise was most helpful. As Walker tried out different arrangements, Joker would provide the reality check: "delay the introduction of this loop," "the feel isn't right here," "delete that loop entirely," and the like.

Eventually, the structure was pretty much complete, and it was time for the tweaks. This mostly fell to Walker, with Joker again providing the reality check. One of Walker's favorite techniques is to choose a snap value of an eighth- or sixteenth-note, and insert just the first eighth- or sixteenth-note of a loop as an intro before the complete loop (see fig. 1). With most rhythm track loops starting off with a kick, this provides an excellent lead-in.

All that remained now was some touching-up. After adding a few volume and pan envelopes, the job was done.

I loved what they did with the tracks, but, hey, that's why they're remixers. continued on page 122



E O

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Big Fish Little Fish



Where have all the independents gone?

BY AL KOOPER

The past few months in the business world have demonstrated a big brother mentality that warrants obvious scrutiny. The *really* big fish have gobbled up the medium-sized fishies in alarmingly large numbers. In the record company domain, Seagrams bought MCA/Universal and AOL acquired Warner Brothers. That was bad enough. At press time, it's rumored that AOL/Warner is about to purchase EMI. Up to now we fretted about Bill Gates owning the world, but it was Steve Case we should have been watching all along.

If this merger comes to be, there will be Sony (the once-American Columbia/Epic Family, now owned by the Japanese), MCA/Universal (which already ate Chess Records and ABC Paramount Records at an earlier meal, and later wolfed down Polygram), RCA (owned by the German Bertelsmann concern), and AOL/Warner (which scarfed up Elektra and Atlantic Records previously, and now threatens to swallow Capitol Records as part of the EMI merger).

And that's it.

All the enterprising independents (Virgin, Slash, etc.) were devoured long ago. Only the rap/hip-hop community boasts some unencumbered independents, and it seems that a few of the most prominent of their CEOs are languishing in prison or cooling their heels out on bail awaiting their days in court. It's unlikely they'll be cashing their Westinghouse

stock in any time soon. In the retail world. it's pretty much more of the same: Guitar Center, Sam Ash, and MARS have forced the mom and pop music stores into unwanted bankruptcies with amazing discount price leveraging. When I was a New York City 12-year-old, I walked into Manny's Music Store on West 48th Street with my dad and he bought me my first electric guitar. I traded with the familyowned Manny's (that's "family" with a small "f") all my life until last year, when Sam Ash abruptly took them over as they sadly floundered; barely able to stay alive in the world of the big fishies. Admittedly, Manny's was a big fish at one time. (Albeit a big fish in a small pond --- management would never expand beyond one

location.) The signed photos that still grace that establishment tell the story of the clientele who once leaned against those hallowed walls: Satchmo, Miles, The Who, Cream, Hendrix, The Stones, Alice Cooper, and ZZ Top to name but a few. But the photos couldn't save that store and they *still can't*. When Manny's was taken over and reopened, the *true* vibe and personality of the store immediately oozed out the mail chute as Sam Ash, who now dominates the real estate on West 48th Street, turned its prized catch into just another minnow in its Manhattan ocean.

Russ Solomon, owner of Tower Records, created a nationwide nightmare for independent record stores. He built incredibly-sized monolithic palaces of retail overflowing with customer choices. He

Up to now we fretted about Bill Gates owning the world, but it was Steve Case we should have been watching all along.

merged video, music, and reading materials into the Coca-Cola of retail record shops. One of his most powerful stores, on the corner of Newbury Street and Massachusetts Avenue in the heart of Boston's Back Bay, was rumored at press time to

have had its expiring lease incredibly outbid by the rival copycat Virgin Megastore. Word on the street is that Tower will have to vacate this plum location by the summer of 2001. This store is such a mainstay of the community, that a Grauman's-style display of famous musical footprints stands outside the entrance in a Solomonsponsored Walk-Of-Fame. What's to become of Aerosmith's cemented tootsies? To the victor go the cement spoils? The mind boggles.

Like Moses, I stand on the mountaintop and call out to you — we are breaking the Ten Commandments of the human-side of business. (Is there even a human aspect of business left?) We are surely headed toward a world where larger-than-life, impeccably groomed Fidel Castros stand at the

helm of each industry, firmly in charge, until Steve Case (or the next Steve Case) comes roaring in on a tidal wave of assets and sweeps each natty Fidel into his everincreasing-ocean-that-will-submerge-allland-as-we-know-it-today.

The handwriting is on the wall and on these pages. (Even *these pages* are involved. The owners of this magazine own six other influential music-industry rags and surely dominate the musicmagazine market.)

I fear it is too late to do anything and that we all will suffer horribly as we gasp for our collective final breaths, drowning in the wake of this inhumane tsunami. I don't know what the answer is. I only know what the cause is: Greed — for money and power — same as it ever was.

Power Tools For Your Ears...



Headphones are critical to the success of your recording project. Years of creative effort, thousands of dollars in studio time or equipment, and the future of your professional music career depends upon the stereo mix that you hear prior to mastering.

beyerdynamic headphones are the ultimate audio tool crucial to completing your project. Offering exceptional value at every price point, beyerdynamic's sealed headphones are characterized by:

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Gregg Mann



PHOTO BY ED FREEMAN

The multi-talented studio pro tells how to make the most of a mix

BY MR. BONZAI

Mr. Bonzai: How did you get this mixing job in Vienna?

Gregg Mann: A few years ago, I was invited over by the Austrian Ministry of Fine Art to teach basic engineering skills at a music workshop. I met some new artists, and they later got a big record deal and asked me to come and mix their new album. It was really great going back and

working with them at MG Sound --- great people and a top, well-maintained studio. Do you have a preference for analog or digital?

I'm an old-time analog fan, but I have to admit that the newer digital consoles are sounding much better. The clean, quiet sound is a real plus, and, although the EQ and compressors are where I would usually have reservations, they're actually sounding good. I still prefer maybe I like the hiss. Any special gear that you must take with you to projects in foreign countries? I don't carry that much, really. I usually pre-order some Focusrite, Summit, or Neve outboard gear - it saves me the trouble of lugging gear through customs. When do you know the mix is finished? I know I've got the

mix when I can listen

to the song ten times in a row and not find something that I want to adjust - could be the hihat, or riding the vocal a little more - if nothing stands out, then I know I've got it. But in another sense, the record is never finished, even though it's ready to go

Suspect: Greag Mann, Mannmade Music

Occupation: Engineer, producer, programmer

Birthplace: Washington, DC

Residence: Brooklyn, NY

Vehicle: Isuzu Trooper

Diet: No meat

Identifying Marks: Numerous tattoos: tribal images, pyramids, and the Egyptian Eye of Horus, done by famed tattoo artist Jonathan Shaw

Pet Peeve: "I hate to have to go and look for my assistant engineer."

Selected Credits: Suspect has done live sound, recorded and/or mixed Dionne Warwick, LL Cool J, Jodeci, Ornette Coleman, Gwen Guthrie, Al Green, Black Sheep, Heavy D, Patti LaBelle, Peter Gabriel, Ice-T, and Keith Sweat

Notes: We met in Vienna at MG Sound Studios, where Mann was mixing the Austrian rock and rap group Superherorockstar, featuring guest vocals by the celestial Janice Robinson (who is opening for Tina Turner in concert this spring).

out into the real world and become a CD. Do you ever get tapes that are really bad and you have to fix 'em? Sure — with a bad

recording you can fix it in the mix. A bad performance can be fixed if it's a timing problem, too, and even a pitch problem can be fixed. For me, it's usually a problem with the recording - bad EQ for example, so you have to work on that to bring it to the level of acceptability.

Here's a little trick I have for a bad snare drum recording: I take a feed from the snare into the cue system and take a small speaker, like an Auratone, and place it on top of the snare. Then I mic the

drum from underneath. It gives me something to work with and can make an amazing difference.







World Wide Audio, without leaving your studio.

Rocket Network takes audio production beyond the boundaries of studio walls, making connections that let you work with anyone, anywhere, anytime. It's like a global multi-track, ready around the clock for musicians to lay down parts, voice-talent to deliver lines, or producers to audition mixes. No time zones, no jet lag, just pure audio productivity.

On-line Flexibility.

Rocket Network[™] uses the Internet to allow professionals to work together on audio productions without having to be in the same physical space. Instead of shipping tapes from place to place or renting high-capacity phone lines, you log into your Internet Recording Studio, where Rocket Network handles the details of passing your parts to others and vice versa. That leaves you free to concentrate on capturing the perfect take, using your own local system to record and edit. Whenever you're ready for others to hear your audio or MIDI parts, you simply post your work to the Internet Recording Studio, automatically updating everyone else's session.

Full Audio Fidelity.

With Rocket Network, there's no compromise in audio quality—the system handles files in a vast range of formats and compression levels, all the way up to uncompressed 24 bit/96kHz. And you don't need access to a super-fast connection; DSL or T1 is great, but you can also work productively over a humble 28.8 dial-up. The system supports multiple user-defined presets for posting and receiving, and handles all conversions, letting everyone participate in their own preferred format. That means you can conduct a session in a speedy, low bit-rate "draft" mode, then move on while the final parts are posted in the background at full-fidelity.

Professional Tools.

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All rights reserved © Rocket Network, Inc. 2000. All other product and company names are TM or ® of their respective holders. CIRCLE 46 ON FREE INFO CARD Where did you learn the fine art of engineering?

I learned audio at Media Sound Recording Studios in New York — that's where I cut my teeth. It was the number one studio in New York at the time and I got a great start by working with engineers Ron St. Germain, Gary Lyons, and Harvey Goldberg. Could you tell me about your work with Dionne Warwick?

Last summer, I recorded Celia Cruz for

her duet on Dionne's new English/Spanish version of "Do You Know The Way To San Jose?" I met Dionne through Gladys Knight when I was mixing for her in Las Vegas at the MGM Grand.

LL Cool J?

LL is a great guy, down to earth and honest. I recorded "Hot, Hot, Hot" and "Phenomenon." What I brought to the record was the ability to move quickly. LL is so professional, and so fast in the studio — he just hates to wait. You have to be ready for him.

Ornette Coleman?

Ornette is one of the originators of free jazz, and I did three albums with him. It's almost all live recordings with a little programming by his son, who produces the albums. I did tours with Ornette, and he's one of my greatest teachers — opening my mind about mixing and music itself.

He forced me to do something that I



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THE BIG PICTURE



CIRCLE 59 ON FREE INFO CARD



wasn't accustomed to. I normally build a mix starting from the drums and bass, to the melodic instruments, and then to the lead, be it the horn or a vocal. He asked me to start with the horn and build backward, bringing in the drums last. I had never mixed a record starting with the main instrument first. He forced me to stretch my technique and reverse my whole way of thinking. You have to plan ahead, and it benefited me in the end — now I have no rigid rules for mixing.

AI Green?

He was coming back from one of his religious hiatuses, and the vocals were pretty much done, but it was an honor to work with him as both producer and mix engineer.

Curtis Mayfield?

I was working with a group that Gary Katz had signed, and they were asked to record "Let's Do It Again" for a tribute album. I brainstormed an intro that worked like a skit and they loved it. They asked if they could use it to open up the album. Peter Gabriel?

I did a remix of "Steam" for Peter. Hank Shockley from Public Enemy in New York was asked to do a remix for Peter, and Hank knew my work, so we met up at Music Palace for the remix. Ice-T?

We go way back to when he was working with Africa Islam, and the Zulu Nation. I did "409," which was a track for his first album. It's about the household cleaner 409 — this was when people were very meticulous about their sneakers. Ice-T wrote this song about how he kept his shoes clean. I mixed the record for him — I had known him from the days when he used to hang out at the Apollo when I was the house engineer. Of all the work you've done, what has achieved the greatest commercial success?

The record I mixed for Jodeci, "Lately," was their biggest selling single and went platinum. It got me a lot of recognition. I mixed "Ain't Nothing Going On But The Rent" for Gwen Guthrie, which was a classic song that got a lot of praise. Ornette Coleman's *Sound Museum* won a *Downbeat* award for Jazz Album of the Year. In the hip-hop world, "The Choice Is Yours" for Black Sheep got me a lot of recognition and lots of work.

What's coming out next?

In the past year I've been working a lot in Europe. Keziah Jones is from London, and I produced and engineered some songs for him — he's a rock 'n' roll blues artist doing really well. I think the *continued on page 122*





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> CIRCLE 30 ON FREE INFO CARD World Radio History

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FREE INFO CARD

Keeping his Edge

Producer-engineer Jerry Finn reveals his secrets for mixing today's hottest rock acts

BY BOBBY OWSINSKI

Jerry Finn is one of the industry's new breed of engineer/producers — brought up in the techniques of his successful predecessors, yet willing to adapt those methods to fit today's music and artists. Finn had his mixing debut on Green Day's Dookie, and then did their follow-

up, Insomniac. Next, he went on to produce and mix Rancid's Out Come The Wolves and Life Won't Wait, plus he worked with the Presidents of the United States Of America, Goo Goo Dolls, and Beck. Jerry Finn adds a distinctive edge loved by artists and listeners alike. In an excerpt from my upcoming book Mixdown, Jerry tells us the secrets he uses while mixing on the wild side of rock 'n' roll.

EQ: You still do many records with relatively small budgets. How much of the money is spent on mixing in those cases? Jerry Finn: The majority. A lot of times when I get called in at the end to do a record like that, my mix budget ends up being at least twice as much as the budget for the rest of the album.

My manager and I always try to work it out with bands that have smaller budgets, though. I've done a lot of indie stuff with bands that were my friends for anywhere from free to half my rate just because I love the music.

Do you usually have to work fast because of the budget? Not usually. I generally take about 10 to 12 days to mix a record. Some take less; some take more. I think we did *Dookie* in nine days. *Insomniac* took 11 days.

I mixed Beck for a PBS show called Sessions at West 54th. We were supposed to mix only four songs in one day, and it went so well that we ended up mixing seven songs in ten hours and it came out great. The stuff was recorded really well, and his band had actually just gotten off a year and a half tour, so they just nailed it. It didn't really require any fixing. And Beck is someone who really trusts his instincts — he doesn't sit there second-guessing himself. We just went straight for what sounded right, and just nailed it.

Before you start a mix, can you hear the final product in your head?

Yeah, that's one of the requirements for me to feel comfortable going into a

record. When I'm sent rough mixes, I really need to hear where I would take it in order to feel comfortable. Sometimes the band tells you what they want and the producer tells you what he wants and the A&R guy tells you what he wants — and they're all completely different things. That can be a bit frightening because you end up being the punching bag for their arguments. [Laughs.] I usually can hear the final mastered record from day one, and then it's just trying to get that sound in my head to come out of the speakers. Do you start your mix with the kick drum, overheads...?

Just out of habit, I probably start at the far left of the console with the kick and start working my way across. Lately, I've tried to put the vocal in early in order to



THE NEW GUARD: Jerry Finn's expertise can be heard on Rancid's ... And Out Come The Wolves. Finn produced and mixed the album.

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create the mix more around that. In a lot of the punk rock stuff you get the track slamming and then you just sort of drop the vocal on top. But, for the poppier stuff, I've found that approach doesn't work as well because the vocal really needs to sell the song. So I've been trying to discipline myself to put the vocal up early on before I even have the bass and guitars in, and then kind of carve those around the vocals.

One thing that I do with drums, though, is try to get the room mics in early on before I start adding reverbs and stuff like that to the snare. I try to push up the room mics and get the sound right on those, and I try to provide a lot of the drum ambience naturally without going to digital boxes.

After you put the drums up, where do you usually go from there?

I'll get the drums happening to where they have some ambience, then put the vocal up and get that to where that's sitting right. Then I'll start with the bass to my surprise, I've nailed it and it all falls together, and then other times when I get the guitars in there, they eat up a lot of the ambience on the drums. Most of the bands I work with tend to have several tracks of very distorted guitars, and they want them all real loud, so then I have to go back to the drums and adjust for that.

How do you deal with lots of real big crunchy guitars?

When every guy in the band thinks he's the loudest, that's when I know I've nailed the mix. I've always tried to just make it so that you don't have to fight to hear "I usually can bear the final mastered record from day one, and then it's just trying to get that sound in my bead to come out of the speakers."

create tension to kinda pull you into the next part. But, overall, I try to make it so you can hear everything all the time, and that generally comes through EQ. Like I'll find the bite in the guitar and make sure that the snare isn't also occupying that same range. Then I'll make sure the low end on the guitars doesn't muddy up where the bass is sitting. And I also have to keep the kick and snare really punchy to kind of cut through all the wall of guitars by multing them off, hard compressing and gating them, and sneaking them back in under everything.

Do you find you use compression on many things?

Yeah. I'm a big compressor fan. I think that the sound of modern records today is compression. Every time I try to be a purist and go, "You know, I'm not gonna compress that," the band comes in and goes, "Why isn't that compressed?" Some, yeah, I compress the bus, although I'm very sparing on certain records. *Dookie* for Green Day had no compression on the bus at all, and the Super Drag record that I produced and mixed last year didn't have any either. But if I think it's appropriate for the music, I'll get it on there.

Are you compressing everything else individually as well?

Lately what I've gotten into doing more of is multing it off, like I said. The kick and snare I'll put through maybe a [dbx] 160 and very lightly compress it, maybe pulling down half to 1 dB. Then I'll mult them off and go through a new 160S and really compress those and sneak them up underneath, so you're basically hearing the character of the drum you recorded rather than this bastardized version of it. Then I also send all the dry drum tracks, not the rooms or overheads, but the kick, snare, and toms, through another compressor and sneak that in to give the kit an overall sound. Distorted guitars I don't compress as much because, when you get a Marshall on 10, it's so compressed already that it

ORIGINAL MIX: Finn made his mixing debut on Green Day's breakthrough album, *Dookie*.

and make sure that the kick and the bass are occupying their own territory and not fighting each other. Sometimes, anything. On certain parts of the song, maybe I will bury something a little bit or push something a little louder to



doesn't really need it. But cleaner guitars or acoustic guitars, I'll compress. And I actually got into doing the vocals the same way I do the kick and snare multing it off and compressing it real hard and sneaking that under the original vocal.

When you say "real hard," how much do you mean?

I would say 10 or 12 dB and at a ratio anywhere from like 4:1 to 8:1. My compression technique is something I learned from [engineer-producer] Ed Cherney. He was telling me about compressing the stereo bus when I was assisting him, but I use the same technique on everything. I set the attack as slow as possible and the release as fast as possible so all the transients are getting through and the initial punch is still there, but it releases instantly when the signal drops below threshold. I think that's a lot of the sound of my mixes. It keeps things kinda popping the whole time. Also, you can compress things a little bit more and not have it be as audible.

Do you add effects as you go along, or do you get a mix up and then add them? I'm pretty sparing on effects. Over the last year and a half or two years, I've gradually tried to wean myself off of any digital effects. The last six or so things I mixed, the main vocal effect was a plate reverb and a tape machine, or space echo for real tape slap.

Are you delaying the send to the plate? Depending on the song. Sometimes it works, but with a lot of the music I do, the tempos are so fast that you don't really need to do much delaying because you can't really hear it. It's like the reverb needs to speak right away and then go away. I'm a big fan of the EMT250 on snare. That's probably been a standard since day one on my mixes. Electric guitars tend to stay dry, and bass is always dry.

How much do you use [Yamaha] NS10's, Auratones, or the main studio monitors?

I use the main studio monitors maybe one percent of the mixing time, if that. I know a lot of people say, "Well, I like to go to the bigs and listen to the low end and make sure that's in order," but the big monitors are so inconsistent from studio to studio that I can't trust them.

I like to check my mixes in mono, so I do use Auratones a lot. NS10's are sort of a necessary evil. Most producers and bands that I work with are used to *continued on page 122* Think how hard it would be to own every piece of audio gear you may ever want to use... ever!

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Matter of FAQs

Sweetwater tech support shares and answers their most frequently asked questions

BY DANIEL FISHER

Sweetwater is a direct retailer of hightech musical equipment with a client base of almost 200,000 customers. Our in-house tech support staff handles about 2,000 incoming and outgoing tech support calls from our customers each week. We also get about 100 e-mailed tech questions per week. Here are some of the most frequently asked questions (FAQs) we encounter:

Why doesn't Logic Audio recognize my MOTU MIDI interface?

You must use OMS with Logic Audio in order for Logic to see the MOTU MIDI interface. Here's why: Logic uses OMS to "see" the MOTU drivers that FreeMIDI installs in your extensions folder. Here are the steps needed to make this work:

1. Download the latest versions of OMS and FreeMIDI and the necessary drivers. For OMS: <u>www.opcode.com/</u> <u>downloads</u>. For MOTU: <u>www.motu.com</u>, then click on Downloads.

2. Delete any old versions of OMS and FreeMIDI from your computer.

3. Install OMS and then restart your computer, but *do not* set up or configure OMS!

4. Install FreeMIDI and again

restart your computer, but *do not* set up or configure FreeMIDI!

5. Now make sure everything MIDI is connected and powered up. You can then launch and set up OMS. If you have a USB device, you do not need to check the modem or printer ports in the device search dialog, just leave them blank and hit Continue — OMS will find your MOTU device on the USB bus automatically. Make sure you test your studio with the Test Studio function located under the Options menu in OMS.

6. Now save your OMS Setup and then launch Logic. Logic may or may not detect OMS upon startup. If it does, it will ask if you want to use OMS...say yes! Logic will now boot-up with all of your MIDI devices intact. If it doesn't, select "Use OMS" in the MIDI Communication Preferences in Logic.

7. Finally, disable any FreeMIDI Extensions in your Extensions Manager.



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I need to send my Alesis ADAT XT in for servicing. Is there anything special I need to do to ship it safely?

Yes. The ADAT XT, ADAT XT20, and ADAT LX20 have a Shipping mode in their service menu. This mode positions the transport parts so they are less likely to pop loose in the event of rough handling.

To access this mode, hold Input buttons #1 and #7 when powering up. This puts the ADAT into Test Mode. The first menu item will be "0 SHIP." Press Play to engage Ship mode. The unit will make loading sounds for about five seconds, then "SHIP READy" will appear. Now the ADAT is okay to ship. The unit will return to normal operation the next time it is powered up. Using Shipping mode is not only good for shipping, it's also a great idea for touring acts, as the handling there can be equally rough.

The Alesis M20 does not have this option. Its transport is made differently so it doesn't require a shipping mode. The original (black face) ADAT does not have this option specifically, but a similar effect can be achieved with these steps: Hold #1 and #7 Inputs while powering up. "ProG" will appear. Press the Pitch Up button seven times to show "CAP." (with a period after it). Now press the Autoplay button to remove the period next to the word "CAP." Place a tape cartridge in the loader, but don't slide it in all the way. Instead, hold onto the edge firmly. Press the Pitch Down button and the loader carriage should go down without the tape. Press the Stop button and the loader should move for about 5 seconds, then the stop light should stop blinking. Power off and you're ready to ship. The unit will return to normal operation the next time it is powered up.

Here're some general ADAT packing tips: • Don't ship ADATs inside rack cases! These are not built to prevent the ADAT from twisting and, unless they are heavily shock-mounted, the front panel will bend if the rack case is dropped. Shippers may reject any shipping dam-

age caused this way.

• Do use plenty of packing. Seal the ADAT in a plastic bag. Service centers have to charge for the time needed to clean packing dust and chips from inside the ADAT. Place the ADAT inside the original box (if available) with packing, then place that box inside another with four more inches of packing material around it.

I'm trying to use my Imation SuperDisk drive with my Mac G4 (OS 9) with no success. What am I doing wrong? G4 owners with Imation SuperDisk drives will need to download the latest set of extensions in order to mount floppies and authorize software. The new version 3.2 is a single "stuffed" downloadable file that contains two extensions (USB SuperDisk Driver and USB SuperDisk UT Driver). The USB Floppy Enabler remains unchanged at version 2.0. To find the software, go to <u>www.superdisk.com</u>, then click "Drivers & Software."

Note that, although the Pro Tools 5 CD-ROM updates your existing USB driver to v2.0, Digidesign recommends that you download the newer 3.2 version from the Imation SuperDisk site.

Also, for all OS 9 users with Imation SuperDisk Drives: Do not boot up your system with a disk inserted in the USB drive. Upon startup, OS 9 now checks the USB drive in order to allow booting from a USB device. If a disk is present, the Imation drivers are bypassed and Apple's own drivers are used. If no disk is detected, the OS will use the Imation driver.

I downloaded some Kurzweil samples off the Internet and I'm trying to open them up with AWAVE software (which reads Kurzweil samples). This works fine except when I try to open samples that are split over several floppies. Why is that? When a Kurzweil file is spread over more than one floppy (it's called a Split-File), the samples, header information (sample start, sample end, loop points, tuning, etc.) is stored on the first disk before any of the actual sample data is stored.

The problem is that, although your RAM samples are spread out over several floppies, the crucial sample header information lives on the first floppy. This means that only the samples that fully reside on the first floppy will be able to be loaded by AWAVE.

Your best bet is to borrow a Kurzweil and use it to copy your larger samples to a bigger medium, such as a Zip or Jaz disc. Now the samples are no longer split, and AWAVE will be able to read them.

I'm using a MOTU 2408. How can I stop my MIDI and audio from drifting apart? Unless you're synchronizing to external timecode, most MIDI and digital audio sequencers automatically clock MIDI and audio together. Digital Performer definitely does this. Make sure that you do not have "Slave to External Sync" checked in the Basics menu. If you're sync'ing your 2408 system to ADATs or to a Digital Timepiece, make sure the appropriate connections are made (ADAT sync or control track on the PCI-324), and that you have "Sample-accurate" selected in the Receive Sync window in the Basics menu of Digital Performer. I just purchased a Lexicon Core2. I've installed the software, and everything works properly, but the Lexicon Studio Panel will not launch. Why is that? There is a bug fix on the Lexicon Web site that solves the problem (www.lexicon. com/downloads/core2 current studio_download.htm). This is a problem with Windows 98. As soon as this file is downloaded and executed, the problem goes away.

I have a Digidesign Digi001 connected to my Mac G4. But when I launch Pro Tools LE, an error message appears: "The application Pro Tools could not be launched because the hardware could not be found, or it is being used by another application." Then either my Mac or Pro Tools freezes or I'm prevented from continuing the launch of Pro Tools.

First check the settings in your Mac's Sound control panel. Make sure that neither the Sound Outputs nor the Inputs are set to "Digidesign." Instead, set them to "Built In." It's important to understand that the Digi001 cannot be "shared" or used by two applications at the same time.

If the Sound control panel is set correctly, try checking the 001 card's seating, and verify the 001's cable connections and check for bent or damaged pins in the connectors. You can also verify the Extension set and recommended control panel settings. If everything checks out okay, reinstalling Pro Tools LE has been known to help.

Can you give me the correct button pushes for my ADAT and TASCAM multitracks to determine how much time is on the recorder?

For ADATs (with the exception of the M20), you can check the hours by pushing the Set Locate and Stop buttons at the same time. For the M20, it's the Peak Clear and Stop buttons.

The DA-88 and DA-38 procedure is to hold the Stop and Play buttons on powerup for head-play hours (d.), and Stop and Ffwd on power-up for search hours (d.s.). Cleaning cycles (cle.) should also be considered head-wear. This is found when you enter the machine's Cleaning mode.

Daniel Fisher is the director of Soundware Engineering at Sweetwater in Fort Wayne, Indiana. You can access Sweetwater's Web site, which features tech tips, downloads, demos, and product announcements, at www.sweetwater.com.

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CIRCLE 18 ON FREE INFO CARD World Radio History

The A&R Perspective

Interscope Records veteran Tony Ferguson reveals just what those guys are thinking

BY MICHAEL JAMES

Interscope Records' A&R vet Tony Ferguson has been with America's hottest modern rock label since the beginning. He made his bones by signing and nurturing artists such as No Doubt and Reverend Horton Heat. Accomplished as a manager, producer, sound designer, and musician before he devoted himself to A&R, Tony possesses insight into many facets of the recording process. As guitarist for the '60s UK bands Christy and Unit 4+2, Tony had hits with "Yellow River" and "Concrete and Clay." He paid his dues as a session guitarist until the mid-'70s, when he became established as a producer. After moving to the United States in 1980, Tony honed his management chops as part of the Grand Funk Railroad and Clarence Clemmons organizations. He generously shared his perspectives with me in late summer 1999 in Los Angeles. The following are highlights from our conversation.

EQ: How did you originally connect with Interscope?

Tony Ferguson: I teamed up with a band that just got signed to Geffen Records, called Lone Justice, and that led me to a meeting with a then-very-famous producer named Jimmy Iovine. And Jimmy and I got on. In 1985 he asked me if I wanted to move out to California and set some things up over here and help refurbish A&M studios. We found ourselves with free time. Jerry Moss and Herb Albert pretty much just gave us access to phones and all this office space, so we decided to form a management company. We ended up managing Lone Justice. Jimmy was still making records, even though

he was becoming dissatisfied with the record-making business.

I once heard him describe producing as a young man's sport.

Yeah, he'd had enough of it. To cut a long story short, Jimmy decided to form a label, [which was] to be funded by Atlantic. He met up with an independent financier in Chicago named Ted Field, who had operated in the movie business very successfully and who was also starting a label. Rather than go up against each other, they decided to form a new label. After multiple meetings, they decided to pool their resources, and that's how Interscope started. They invited me to come along, and I have been with Interscope since 1990, almost ten years. That's the background.

With the current state of the record industry, it seems that the opportunity is ripe for beginning recording pros to have a fast track to the top. It's true. Back in the '80s you didn't have the technology available today. A kid can make a very well-made record out of the living room, with ADATs and computers that never existed before. Artists are no longer reliant on record labels to allow them the ability to get into a studio, to make music in a professional manner. Now, with the technology available, any kid that knows his way around a computer and has at least some sense of music can put a piece of music together, and if he has a good ear for what music should sound like, he can do a good job of it.

Do you think a large proportion of new artists on Interscope are project studio recorded instead of commercial studio recorded?

I would say that it is close to 50/50; some are done on ADAT and then remixed or mastered in a commercial studio. By and large, most new artists,



NO DOUBTS: As an A&R man for Interscope records, Tony Ferguson signed No Doubt to the label and nurtured them during the recording of *Tragic Kingdom*.

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World Radio History

TECHNIQUES BUSINESS



TALENT SEEKER: Tupac Shakur was another performer that Ferguson signed and helped develop into a superstar artist.

when they sign with a major label, ask to go into a commercial recording studio to try to emulate the tapes they made in the past — sometimes disastrously.

To get 10 percent more recording quality or better performances? Or do they think there's going to be a miracle by working with recording professionals? There's a myth in artists thinking that a miracle's gonna happen by working with pros. That's over-exaggerated. That's kind of what drives the producer's end of things. We've had many a situation when we spent lots of money trying to duplicate - on a professional level - a tape that was originally sent in by an artist, which was good enough to get them signed. We already liked the vibe of the track; it was done cost-effectively on relatively inexpensive equipment, and we spent hundreds of thousands of dollars trying to get the same sound again.

Particularly with bands, you're going to find it less and less a desire to

go into a [commercial] recording studio. I think with a solo artist, where a team has to be assembled around them — like producers — musicians, orchestrators, arrangers, that's a different ballgame. Then you need a commercial studio.

From the record company's perspective, do you prefer a format like an analog Studer to an ADAT, DA-88, or something on that level?

I've heard records made on ADAT where I couldn't tell the difference.

Do you think it really matters when it comes down to it?

Depends on the individuals.

Do you do much interfacing with recording professionals?

Yes. I'm very particular about the way my records turn out. Mastering is a very critical thing to me, yet many A&R guys don't go there. It blows my mind that labels don't go to the sessions because the difference that mastering can make to a record is immeasurable. I in-

terface on every level because I want my records to sound the best they can. I go to preproduction and recording sessions, but not every day because I need to keep an objective opinion about the record. You have to play the devil's advocate. If you get caught up with the artistry of the artist, you're sunk --- you're no good to the artist. So I'm usually in and out very quickly because I don't want to get bombarded with everything that the artist is dealing with in the studio and lose my objectivity.

What qualities do you look for in a producer or engineer?

Somebody who's sympathetic with the artist. If it's a rock band, somebody who's young, aggressive, and ready to take risks. If it's adult contemporary, someone with a certain level of sophistication in their artistry. Usually with something like that, someone to pay particular attention to the voice, whereas most rock producers don't. They're more about the drums and guitars, and the voice is another part of that instrumentation. Other records are mostly about getting the right vocal sound. It all depends on the kind of record you're making. An A&R person needs to pay attention to the type of record — you

don't want to put a rock producer on a Barbra Streisand record.

It depends on how comfortable the artist and record company are with the producer.

That's true. It's somewhat of a democratic situation. The artist has got to live with the engineer and producer in the studio for hours and hours on end. So they've got to feel comfortable that they're with the right people. And, as an A&R person, the first thing is to make sure that happens and to read between the lines. Sometimes a new artist is afraid to say something for fear of alienating the record label because of some hotshot producer and engineer they brought in. An A&R person's got to recognize that and pull the plug immediately, because you could be throwing good money after bad.

Is it more important for a recording pro to give artists what they want or what they need?

Definitely what the artist needs. You go

World Radio History

with what the artist wants up until the point where the objective opinions become what the artist needs, then you give the needed objectivity. Then you need to take control. You usually find that happens in mixing. Assuming the artist has given a good performance, unless the producer and engineer are on drugs, it's hard to screw it up. The mixing becomes crucial; it's the final dash to the wire. Sometimes the artist and producer, particularly with rock bands, buddy up and listen at high SPL. Everything keeps getting louder more treble, more bass, more snare, more guitar - until you have noise. That's when it's best to talk to the producer and suggest that we bring in somebody with fresh perspective to mix. Someone with clean ears and an objective opinion who can handle this type of music and give them a shot at it. If the team is uncomfortable, you suggest we try just a couple of people on a couple of tracks and see which one fits. Hopefully you get it right.

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Do you master differently for the record than for radio?

Sometimes - particularly if you think you've got a radio hit. You may want to do a special radio edit or compensate for the radio's compression with EQ. You might make it a little brighter, but not too bright, because you have to take into account that different radio stations use different compression ratios, a lot of them very severe. You can actually hear the record "sucking" on the radio. That's why you'll find a lot of people will have the vocal up loud. Great mixers like Tom Lord-Alge mix the vocal loud because it's the first thing that hits the radio's compression, which ultimately brings the track up to surround the vocal. Artists, however, will take the radio mix home and then call me, assuming that the vocal is too loud.

But it works for radio. Many times records are severely compressed before they go to radio. Is that a problem?

I used to think so until David Holman mixed some Bush and No Doubt singles that I thought were overly compressed, but his stuff sounded really cool on the radio. But, again, he had the vocal hot, so it hit the radio compressors in the right combination. As long as the ratio of music versus fidellty versus vocals is in the right ratio for that piece of music for listening on a normal stereo system, it should hit radio at the right compression levels. It amazes me that music is now mixed for radio rather than records. Is it fair to say that today's new music is almost totally radio driven, and that if you don't cater to it you're unlikely to have a successful record?

Absolutely. If you're wedged between two songs on the radio and you don't sound like you fit, that will have a profound effect on the audience. The audience may not know what's happening, but they feel like the record just doesn't come alive. On a creative level, is there anything you might suggest to up-and-coming audio professionals?

Get to know the equipment, keep up to date with it, and get as much hands-on experience as possible. Then it's all about your instincts, understanding and recognizing good music, and associating yourself with that good music. Try and keep the talent bar up to a level that you feel is going to get you noticed. People don't necessarily notice *continued on page 122*



67



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CIRCLE 21 ON FREE INFO CARD



CIRCLE 11 ON FREE INFO CARD

TECHNIQUES RECORDING

speak the language of the artists you're recording. That will go a long way in helping get this sort of gig. If you can read music at all, ask for a copy and make some notes. At the very least, try to listen to a similar recording in advance of the recording date. Not all classical music is the same, just as there are different types of rock music. Learn the turf on which you're playing and you'll be way ahead.

4. No overdubs, punch-ins, or other multitrack tricks allowed. You may be in a close-miking situation, but don't expect to roll back and punch in a correction. Even if you have every instrument isolated in its own booth, the musicians will still be used to cueing from the conductor and each other. Instead, ask for a retake from a certain bar (see why it's nice to be able to read the score?) and, in postproduction, splice the two takes together. Just don't move mics between takes. The sound will never match.

5. Know your tools. There's simply no getting around it. If you're doing a rock gig you probably know what mics to use: Put a '57 on the snare, '421's on the toms, and a D112 on the kick. Then put up '58's on the vocals, some '57's on the guitar speakers, a DI on the bass guitar, and you're done. Sound familiar? I thought so! Just as you learned which mics to use for rock bands, you'll have to learn which mics to use on acoustic classical instruments.

To start, you can forget about using Shure SM58's on most classical instruments. It's just not the proper mic. While you don't have to have a cabinet full of Neumann mics to record classical, you do need at least a few good mics and the know-how to use them. I like large-capsule condenser mics from Audio-Technica, AKG, GT Electronics, and Event, among others. If you're doing a purist stereo recording, then you only need two excellent mics placed in the sweet spot. For most of my classical recordings I'll use a center pair on a common stand in an ORTF array (more on this next month) and "edge" mics on the corners. These will be anywhere from 8 to 12 feet high, depending on the situation. Then I'll add in a few spot mics for soloists or to add definition to the mix. I find that I can do a nice classical minimalist recording with an 8-track recorder and that many mics.

6. Compression is a dirty word. In gen-

eral, there's very little crossover between classical and rock recording and production, especially where processing is concerned. Classical music is supposed to have dynamics. Sometimes you have to compress, but, in a perfect world, all classical recordings have full dynamics. Compress or limit where you anything you do to limit dynamics is really adding distortion. This is where knowledge of the particular genre comes in. For instance. Ian Anderson (Jethro Tull) may need lots of dynamic control when playing with an electric rock group, but your normal classical flutist does not.

Also, you shouldn't need much equalization on a classical instrument. You just need to pick the right microphone and place it properly. It's all about nuances in classical stuff. Anything outrageous probably won't work. 7. When to get close. Up to this point we've been talking about recording only, not sound reinforcement. That's a whole other gig entirely. For the last four years, I've worked with the Maryland Symphony Orchestra for their 4th of July concert at the Antietam Battlefield. We need to cover up to 40,000 listeners located as much as a quarter-mile from the stage. How do we do it? Basically, with a big console and lots of mics. We'll use mics on tripod stands with booms and position them as close as we dare to the instruments. For violins and violas, this could be AKG C535's placed about one foot above them. Cellos get Sennheiser MD-421's, and percussion instruments such as tympanis use ElectroVoice RE-20's or AKG D112's. Horns generally get dynamic mics (you can use your SM58's if they're not beat up), and harp gets a Barcus Berry piano pickup. Then we load up a Yamaha PM4000 console, put up ten delay towers, and tweak everything to within an inch of its life. This is really the only way to do it for live reinforcement.

If you're trying to combine classical and electric instruments, you may need to mic even closer to get rid of the electric "wash." In that case, putting small condenser mics on each instrument is the only way to fly. I like the little clip-on Audio-Technica and Crown GLM mics for this sort of thing. Yes, it's a lot of mics. Yes, it's a lot of channels. But that's how you can do a band like the Moody Blues or Metallica with a symphony.

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CIRCLE 16 ON FREE INFO CARD

Unconventional remixer Dany Saber follows his intuition to please his diverse client list

Story by Howard Massey Firston by Edward Colver-

he worlds of dance, trance, electronica, and techno may be anchored in New York and London, but there's a whole lot of remixing going on in L.A. as well, and Danny Saber is one of the central figures in that scene. An accomplished guitarist, Saber first made his mark in the mid-'90s as a member of the eclectic group Black Grape. His unorthodox techniques soon gained him a reputation as a remixer to be reckoned with, and before long he was deconstructing and rebuilding tracks for a diverse group of artists ranging from Public Enemy to Megadeth; from David Bowie to Marilyn Manson; and from Madonna to Sheryl Crow. In 1997, word of his talents spread to the Rolling Stones, who brought him onboard for their Bridges To Babylon album (Saber co-produced the track "Gunface" and played bass and clavinet on the hit single "Out Of Control"). We caught up with him recently at L.A.'s fabled Record Plant studios as he was completing some tracks done in collaboration with the late Michael Hutchence. Though his conversation is liberally sprinkled with four letter words. Saber came across as thoughtful, intelligent, and sensitive — and as someone who has clearly not lost touch with his street roots.

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EQ: You're a musician who became a producer/engineer, not the other way around.

Danny Saber: I'm an engineer by default. I'm the same guy as most of your readers who are doing sh*t in their house — that's how I started. I had a TASCAM 8-track and one [Roland] S50 sampler and a Juno 106 [synth] and a D-50 [synth]. Then I got an Atari 1040 and I was running [C-Lab] Creator on it, then I got an [Akai] S1000 [sampler]. So, as the technology became more affordable, I just gradually got stuff. I was also fortunate in that I was in the right place at the right time, because, when I started, there was only one sampler — the S50 that was affordable. Before that, you were looking

at a Fairlight. Now it's insane — what do you buy? There's so much stuff that people can use it must be really confusing for kids that don't know what to get.

The great mystique is that, if you have great gear, you don't need great skills — that you'll just turn the equipment on and it will make a great record.

That's all a bunch of bullsh*t. Not to sound egotistical, but the reason I'm doing well is because it's all about content now. Quality sound doesn't mean anything anymore, because anybody can get it. I'm a big fan of all those Steely Dan records — the records you put on not only for



the writing and musicianship and performance, but [also] for the way they sounded. Sonically, those records are unbelievable, and it was really hard to make a record that sounded that good back then. Now, with the digital technology, anybody can do that — I'm talking purely from a technical basis, not from content, because obviously the musicianship and the songwriting and all that, nobody else can do. But it is possible to make really bad recordings with really good gear.

You're right, but my point is that you can get a box now that gives you 800 drum sounds that are all unbelievable.

However, if you overload the console when you

record them, they're still going to sound bad. That's common sense stuff, though. I don't really think there's any trick to that. You either hear that it sounds like sh*t or you don't. And, sometimes,

notinet

things that sound sh*tty are good. What are the common mistakes that people are making in their recordings today?

I think the worst mistake people make is trying to do too much. That's a mistake I've made, and I've learned from it. When you've got something good and it's happening, you don't need to f*ck with it anymore; leave it alone. That's the thing I'm really working on now: I'll spend more time doing something and do less to it. I'm being more patient,

> doing more listening. Not just doing sh*t to do it everything has to serve a purpose, to make the song better or make the performance better.

It comes down to striking a balance. If you don't flesh out a song sufficiently, it comes out sounding like a raw demo, though maybe some songs require that. How do you know when you're doing too much and how do you know when you're doing too little? It's all instinctive; it comes from experience and knowing what feels right when you're doing it. The way I work is totally unconscious - even on a technical level. I don't really sit down and think, "This is what I'm going to do to

this." I just do it. The thing I think I want to try may not be the right thing, but it will lead me to the place where the thing needs to be.

Instinct obviously plays a huge role because, at the end of the day, the purpose of music is to invoke emotion, but you also have to have some technical skills to make a great record.

Technically, what you need to know is how hard to hit the stereo bus, you need to know when you're putting too much compression on something, and you need to know how to EQ to create space. I didn't learn these things the way a traditional engineer learns them. I learned them by trying stuff and then picking things up from other engineers and asking people. Like compression — that's always a really, really dicey area.

So how do you know when something's over-compressed?

When the life is sucked out of it. My idea is to be conservative on those types of things when recording and save it for the mix, because you can always add more, but you can't take it off. If you over-compress sh*t, it'll suck the life out of stuff, especially when you're playing guitars or bass — it'll totally change your performance. It's a subtle little thing, but it can really ruin it. That gets back to what I was

saying earlier: Be purposeful in your decisions. How do you know when to use a compressor and

when not to use one? I think you should almost always start with one and see if it feels right. Again, it's all about instinct, and the way that you learn about these things is by doing them. It's all about trial and error; there's no magic secret trick that you can learn that's going to sort everything out for you. It's all about experience, and all the great engineers are very experienced.

If you're an experienced musician and you come in to play on a track, you hear the chord progression and whatever the vibe of the song is, and you know there are different ways you can approach it. It's the same with engineering; it's like being in any situation in life. When you've been in that situation before, if you have half a brain, vou're able to deal with it a lot better the next time it comes around. So the key for young people starting out is to make the most of every second while they're working. That way, when they get into a situation where it's important and it really means something, they're ready for it. That's the thing that I've noticed with a lot of people in life; they don't take the little sh*t seriously and then something a little more important comes along and they're not ready to deal with it. They're out of their depth, and maybe they could have been a little more prepared if they had worked harder on things that at the time didn't seem very important.

When I was learning, every time I got an opportunity — it didn't matter what it was — I made the most of it and learned as much as I could. And all of that stuff came back around later in a situation with the Stones or somebody like that, where you're really on your toes and you've got to use every brain cell you've got left! [Laughs.] That's what prepares you — the little things leading up to that. Everything goes into everything.



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Now it's totally different. You've got a bunch of kids in studios all over the place in their houses and they're throwing their ideas down. And almost every record I ever made started with that demo. I got all the stuff they did, I took it, and I rebuilt the track around that. Now everything may have gotten sh*tcanned in the process, but that gives you the option. That's what's really important, and that's the advantage now. When you throw down your initial idea and that spark of inspiration's there, you don't have to sacrifice that anymore.

A big fear of young bands is producers coming in and changing everything. There's no excuse for that anymore, because, even if there's one thing on

the tape that's the essence of what that song's about - it could be a loop, it could be some vocal that somebody did in their house that maybe doesn't sound so great technically, but the vibe of it is awesome there's no reason to get rid of that. You redo it, you try and beat it, but if you don't, you've got it. That's the real beauty of the way technology has changed. It gives you options - you've got so many more options those days then you used to have.

Someone in their own project studio theoretically can spend any amount of time on a recording. How do you know when to stop? You've got to know when you're past



the point of positive input. Again, that's something you learn by experience; there's no one defined answer to that. When you're doing a record, there's always this one song that's a bitch that you have to redo 20 times. And then there's the others that, you do them once, and they're awesome. Not only is that a problem for the guy in the project studio, I think it's an even bigger problem for the successful multiplatinum artist who all of a sudden is in a situation where he's got power. Those are the guys who shoot themselves in the foot more than anybody! I've seen it so many times, with singers especially, where the sh*t's happening and it sounds great, but, because of their insecurity or fear or paranoia, they've got to keep beating the thing into the ground until nobody can stand listening to it anymore. And if you do say "enough," sometimes you get fired! Personally, I'd rather walk, and I've done it — with big people, too. I've walked because they just beat the sh*t out of the song to the point where it just wasn't fun anymore.

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So I guess a way you could tell when you're done is when you're not getting off on listening to it anymore. Maybe that's a point to stop and move onto something else and later come back to it with a fresh head. But you always have to remember how you felt when you first came up with the sh*t, because you've got to listen to stuff over and over again — that's the nature of it. And if the excitement still isn't there,

> you've got to differentiate between whether it's because you've heard it ten million times or because it's not good. That's a hard line to cross. Ultimately it comes down to confidence in vourself and not getting too precious about stuff and worrying. There's things I always want to keep working on, but, at a certain point, I just let it go.

Is it always helpful to bring in a second set of ears?

Definitely. It's nice to play finished stuff to people — or stuff that's almost finished — and say, "What do you think?" But you've got to be really careful with that, too, because half the time they're going to tell you it's great

anyway. So you get this sense — it's not really so much what people say to you when you're playing them stuff, but you can look at them and tell if they're feeling it or not. That's another thing you just develop by being in those situations.

I used to go see all these A&R guys when I was nobody — somehow I got a meeting with them and they fast-forwarded through my songs, listening to ten seconds of a song and then fast-forwarding to the next one. At the time, that seemed really degrading and it sucked, but it really aids me now, because I can tell how people are reacting. When you're playing stuff to people, you can tell the vibe in the room if you're tapped into it. But, ultimately, it comes down to yourself; you can't rely on anybody else to tell you whether what you're doing is good or not, aside from the people you're intimately working with. Once you go outside that circle, all you're doing is inviting trouble.

How do you deal with situations where you're convinced that something is right and the artist lsn't?

It depends on the artist and how genume they are in their reasons. Sometimes the artist is saying sh*t because of their ego. But if they're genuinely committed to making the record and taking the musical journey, then you've got to sit there and listen to everything they say and take a hard look at what's on tape and be open.

When I'm not working, I've always got records on because I just get off on listening to music - that's my life. That's what you've got to look for, both as a producer and as an artist: people that are for real and are genuinely on that musical journey of life. But if you're a producer, you've got to listen, because the artist has to be happy - it's their record. So you'll defer to an artist if they insist that their way is right? Well, I won't do it just because they say it's right. But I'll listen to what they say, and we'll talk about it. And, hopefully, in a good situation, we'll mutually find the place that it needs to go. And the middle ground doesn't always have to be a compromise, because sometimes compromise isn't good in music - there's got to be some friction. The best music has always come out of frictional situations. Take Mick and Keith - they may not agree on certain sh*t, but they still make great records. So it's not smooth all the time. But if I really believe something's right, I'll explain it to an artist and I'll tell them why. Ultimately, the final decision ... well, it depends on who it is. But I really haven't had a lot of problems like that. What's the single most important piece of gear in the project studio?

What you need these days is something to get your songs recorded with. If you're doing, say, dance music, it's imperative that you have some sort of sampler; something that you can do beats on and still be able to record some sort of vocal. People are doing instrumental music now, too, so you can do all sample-based records with just a sequencer and a sampler and run it right to DAT. But the way I started was more helping other people get their songs on tape. If you want to go that route, you need something like a DA-88 or a hard-disk recorder — something that can capture whatever it is you need to record.

I think the mixer side of things is less of an issue at first. But, again, it depends on what it is you're trying to accomplish. Do you want to be a producer? Do you want to just record songs? Do you want to make demos? Do you want to make records in your house? Do you want to do underground-dance-music-type stuff? There's a million options, so you've got to go look at what's out there and figure out where you want to start. Ultimately, you may end up somewhere you never thought you'd be — that's what happened to me. I just started out wanting to play guitar, and now look at what I'm doing! [*Laughs*.]

If you can pull it off, you should get some kind of mixer, buy a hard-disk recorder like a Pro Tools/Logic situation, get a sampler and a couple of keyboards, and you're rocking. Then you start adding things, like different guitars for different situations. Only a few years ago, I had just two guitars, and they worked for everything, because that's all I had and they had to. You get what feels right and you can go far with stuff, but I also think it's *continued on page 128*

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CIRCLE OF ON FREE INFO CARD

Forducer Mike Hedges discusses his inherited Abbey Road console, as well as how to get killer drum sounds

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story by Howard Massey photos by Simon Camper

Q went across the Atlantic to speak with producer Mike Hedres as 'ne awaited the start of an orchestral date at Abbey Road Studios. Hedges started as an engineer in the late 1970s, at the height of punk mania. Since then, he has gone on to work with an eclectic group of artists that includes Manic Street Preachers, The Cure, The English Beat, Marc Almond, Alex Harvey, and Siouxsie and The Banshees. His studio in France boasts a unique acoustic design described here in detail — as well as the legendary mixing console used for the Beatles's Abbey Road album. Though tall — even imposing — in stature, Hedges is soft-spoken and thoughtful, the kind of producer you imagine would act as a sea of calm in even the most volatile studio situations. on, then you're halfway there. Then, as soon as you start glossing it up, you're three quarters of the way there.

That approach can also inspire the players to work a little bit harder when they're doing overduls.

Definitely. Also, you get an atmosphere from the track early on, so that when you start replacing guitars and basses and things, you can always go back to that and say, "Hang on, are we losing something here? The bass sounds better, sure, but has it got what that live take had?" And if it hasn't got what that live take had, then you Pro Tools off the live take and fix the bit that's wrong: "That bit's late, pull it a tiny bit earlier." Use it to analyze which are the best bits and keep those.

Do you tend to use a lot of mics on drums, or do you take a more basic approach?

Both. Sometimes we use two [mics] — bass drum and overhead — or even just one overhead in mono, compressed up. And sometimes we can have 12, 14 mics on the kit. It really depends on the song, the band, the drum kit. A fantastic sounding drum kit hit right and hit in the right places can sound good with two or three mics. We often use two mics — one overhead and one on the bass drum — and then move the overhead mic to get the sweet spot. You tend to use quite a lot of compression when you do that, so you have that sort of beefy, roomy, John Bonham sound.

How do you ware for that sweet spot with the one overhead mic?

You do it with your ears. You walk around the drum kit, listen to where you hear the power of the bass drum, and put the bass drum mic there. Then you listen upstairs and move the overhead an inch, two inches, five inches, until you get it right. It's as simple as that.

Would the bass drum mic be very close in, or would it be a foot or two back?

Not very close in; you generally have the bass drum mic nearly as far away as the snare mic would be. So you have it out in front of the bass drum, and, if you want a bit more crack on the sound, you'll actually be moving it in. There



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are so many ways of miking up a drum kit. When I was engineering, I was taught what we used to call the Glyn Johns's way, which was three mics on a kit - bass drum and two overheads both two-and-a-half stick's length from the snare. One is over the floor tom, looking toward the snare, and the other one's directly over the center of the snare. Then you have to adjust one of them for phase. And the bass drum mic is just in front of the bass drum, not in it. After awhile, we started getting quite radical and started putting the bass drum mic in the bass drum, next to the skin. Also, at the same time, I was working with engineers that adopted the '70s way of having top and bottom mics on every single drum.

"Someone who can sing a song perfectly in tune and perfectly in time doesn't necessarily have what it takes to be a great vocalist."

The same thing with guitars — I almost always put a [Shure] SM57 right in close to the speakers. Depending where you have it — outer speaker or inner speaker — you get the difference in tone from the edge of the speaker and the center of the cone. I use a Sennheiser MKH 40 the same way — use it in close with the '57 or back it up for a bit more room. Because the guitars are in boxes — they aren't just cupboards, they're designed so you don't get too much phase problems and things like that you can't really get the furthest mic more than two or three feet away from the speaker while recording backing tracks. I tend to use two amps as well.

If it's a clean sound you're after, you can set up one amp with the clean sound you want and then overdrive another amp in a cupboard with the doors closed, and record that as well. Sometimes we even use three amps, depending on the song. When you record that, you've got to multitrack so you've got two or three tracks of guitar: one clean, one medium — say, half-driven — and then one really driven. As the song progresses, you might use the nice clean track during the verse. As you are coming to the bridge, you fade in the heavier guitar sound then back it off a bit; into the chorus with everything full on, then back to the next verse and drop It all out. It's all done on one guitar track, so it doesn't sound like you've done ten guitar overdubs. It has a different quality; It sounds like a live performance, but you've got real dynamics in the sounds. It's a very effective technique.

You can take the same approach with drum kits. If you're using close mics as well as room mics on the drum kit, you don't have to have both in all the time. If you're coming to the chorus, you can fade the ambience into your drums at the same time you're fading the crunch into the guitars. The effect is like you're just swooping up into the chorus, with real power and filling out. Then you just cut them both at the end of it, when it goes back to the next verse, and it goes *voom*! You've suddenly got this wall of sound cut into a vacuum.

Do you typically record a bass amp track or do you just use the DI?

I always use both, full range. The advantage of having both the bass DI and amp is that it gives you a lot of control over the space the bass is occupying, where it fits into the song. Obviously, the DIs got no life to it — it's a dead, close-up quite "Motowny" sound, which is pleasant. It's one of my favorite sort of sounds, but if you have the amp as well, you have two distinct sounds that you can phase against each other. If you put the DI out of phase, for example, and EQ all the middle into it and take all the bass and top out, and then fade that in, out of phase with the amp, you've got a completely different bass sound. Or you can do it the other way round. It's amazing — when you have a bass amp track that's a bit woofy and it's taking up too much space, *continued on page 126*



by Mitch Gallagher



hese days the delivery format of choice for many studios is CD-R. DAT still has a strong hold with some, but I have to admit: other than for loading a few older tracks into my DAW recently for re-mastering, I haven't even turned on my studio's Panasonic SV-3700 in quite some time. In most situations, CD-R, whether created with a stand-alone CD burner or a computer-based CD-R drive, works pretty well. But, in general, stand-alone burners don't offer much in the way of track-editing capabilities, DSP, re-ordering tracks, and so on — you're pretty much stuck with what you send into the machine. A CD-R drive gives you a lot of flexibility, but requires software, a computer, and often computer-based audio 1/O to effectively make it work; it's less convenient for burning a quick disc now and then. And what about

ML-9600 High-Resolution Master Disk Recorder

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CURSOR CURSOR LEFT RIGHT	TRACK MOVE TRACK START TRACK END UTILITY	SCAN

the need for a way to record and deliver high-resolution stereo masters? The options have been limited: CD-ROMs or a few recorders capable of 24-bit recording have largely been the extent of it.

With the MasterLink ML-9600 high-resolution master disc recorder, Alesis makes a bid to solve the current solutions' limitations on flexibility, power, ease of use, and support for high-res audio. MasterLink integrates a stereo hard-disk recorder with support for resolutions up to 24bit/96 kHz, DSP-processing and track-editing features, and a CD burner into one convenient, tworack unit. As a bonus. MasterLink introduces a new format, CD24, which provides a means for delivering high-resolution stereo files on a CD-R disc (see sidebar "The CD24 Format").

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- 2 Prizes Twelve [12] Grand Prize Winners will receive \$2,000 in cash 2 Prizes Tweive 112] Grand Prize Winners will receive \$2,000 in cash, \$5,000 in Yamaha project studio equipment, and a \$5,000 advance from EMI Music Publishing. One (1) Grand Prize Winner will receive \$20,000 for the "Song of the Year" courtesy of Maxell. Thirty-six (36) Finalists will receive \$1,000. Sevenity-two (72) Runners-up will receive \$100 ght contineates from Guilar Center Stores. 3 Contest is open to amateur and professional songwriters. Employees of JLSC, their families, subsidiaries, and affiliates are not eligible. 4 Winners will be chosen by a select panel of judges comprised of noted songwriters, producers and music industry professionals. Songe will be undered hased upon method, composition and haries.
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STEINBERG **WAVELAB 3.0**

BY CRAIG ANDERTON

Steinberg's WaveLab has gone way beyond its version 1.0 shakiness to become a premier digital audio editing program for Windows. Version 3.0 adds several small tweaks that make operation more efficient or capable, but the biggest addition is the *audio montage*, a multitrack editing environment optimized for assembling bits of audio (called *clips*). We'll cover that soon, but, first, let's discuss the little things.

TWEAKS

Finally, you can do precision pencil waveform re-draws — I've used it several times to draw out the occasional click, with perfect results. The recording section has been completely rewritten, with new options such as "start on sound" and "stop on silence."

Another bonus is that WaveLab can now save in PARIS (PAF) file format. (This is actually of interest to anyone using Mac and Windows, as the PAF format is inherently crossplatform.) And speaking of file formats, in addition to WAV and AIFF, you can now export as MP3 with several different levels of quality using either fixed- or variablerate encoding. You can also save in any

FIGURE 1: The Audio Montage window, being used to assemble a drum loop from individual samples. Note the info line toward the bottom of the window that shows what actions invoke particular functions. In the background is the very useful spectrum analyzer; the foreground shows the new VST Dynamics plugin. And peeking out from the very back is a WAV file that's part of the montage.

> The latest version consolidates past improvements with new features, and comes up with a winner



file format supported by Microsoft's ACM software technology. Furthermore, it's now easy to convert a file to common sample rates from 11 kHz to 96 kHz, or anything in between, at 8-bit to 32-bit floating-point resolution.

A jog/shuttle option "slides" audio past a fixed point, like rocking tape across a playback head. Unfortunately, you can't place the cursor at the final point where the "tape" ends up. I'd prefer it if you could grab the cursor and scrub, because once you found the right spot, the cursor would already be where you wanted it.

Version 3.01 also works out some compatibility issues. Versions prior to and including Version 3.00 could not recognize IDE CD-R drives; that's been fixed with an update available from the Steinberg Web site. Also, I experienced a file incompatibility glitch with Sound Forge when using 3.00, which has also been fixed. Finally, I could never get WaveLab 2.0 to extract data from audio CDs. Version 3.01 works like a champ for anything involving CDs, whether you're extracting or burning.

You also get some new toys. A VST Dynamics plug-in offers frequencysensitive gating, real-time compression (the previous WaveLab compressor, which is still available, only worked offline), and "soft clip" limiting.

Graphically, you'll definitely need a big video monitor; it's just too easy to have lots of windows and views open. Another issue is a certain lack of consistency in the look. This is a minor point, as it doesn't affect performance, but it makes WaveLab feel more like a collection of modules than the tightly integrated program it really is.

THE AUDIO MONTAGE

Opening a montage provides a selfcontained environment (see fig. 1) with two main panes. The lower one supports multiple mono or stereo tracks (they must have a consistent



MANUFACTURER: Steinberg, 21354 Nordhoff St. #110, Chatsworth, CA 91311. Tel: 818-678-5100. Web:

APPLICATION: Windows digital audio editing and processing software.

SUMMARY: Thanks to many small upgrades and one major addition, Version 3.0 takes WaveLab into the heavyweight division of digital audio software.

STRENGTHS: Integrates CD burning, audio extraction, sampler, and 24/96 support. Montage feature is a godsend on some projects. Fast operation. SMPTE sync. Stability far exceeds previous versions. Several user interface tweaks streamline program operation. Accepts DirectX, VST, and WaveLab plug-ins. Can export as MP3 and various compressed file formats.

WEAKNESSES: Doesn't work with AVI files. Allows only two channels of audio I/O (more would be nice for the montage feature). Limited, albeit high-quality, selection of on-board processors. Occasionally cluttered feet due to interface design.

SYSTEM REQUIREMENTS: Pentium running at 166 MHz, 32 MB RAM, IDE hard drive, Windows 95/98/2000/NT (Service Pack 5 required with NT and a PIII processor), MME-compliant sound card, supported CD-R or CD-RW burner.

PRICE: \$599

EQ FREE LIT. #: 102

sample rate, however), and you can add parts just by dragging and dropping. The upper pane provides various views. My favorite is the 60stage spectrum analyzer, but this pane also provides an overview for "express navigating" (click on the overview, and the lower pane jumps to that point), zoom function for individual clips, clip grouping (groups clips together when you want to insert a clip, to ensure that all desired clips move together), edit history, snapshots, markers, a notepad, burn montage to CD, etc. The metering is impressive; not only can you display peak and average VU, you can edit the meter characteristics and display phase info.

The montage is great for building up complex sound effects from constituent parts or layering sounds to create big sounds. As the program can now sync to MTC, this makes the montage almost ideal for audiofor-video applications — assemble dialog, sound effects, and music all in one place. The reason I say "almost" ideal is that you can't import AVI files so you can have a video for reference; you'll need to follow a video device that generates timecode.

A less obvious montage application is creating drum loops by dragging individual samples into the montage, then snapping them to measure/note boundaries. Snapping is not as developed as, say, Cubase; when dealing with bars/beats, it can be tedious to snap clips perfectly in place. Another application is doing drum loop enhancement. Want to add a little extra "oomph" behind the kick with another kick sample? Just bring the new kick into the montage on a separate track.

There are a ton of montage features — duplicate clips, copy them, move, overlap (a track is "polyphonic," and can play back multiple, different clips), or butt up two clips to create

an automatic crossfade with your choice of characteristics. You can cut and erase sections, pull out sections into new clips, and so on...And on...And on....

Another cool feature is that each clip is divided up into multiple "mouse zones," which simplifies editing chores. For example, hold the mouse over the move zone, and clicking/dragging the mouse moves the clip. But hold the mouse over a trim zone, and dragging the mouse trims the clip length. Although there are a lot of mouse-zone options depending on whether you're rightclicking, left-clicking, or using a modifier key, an ingenious info line always reminds you what the options are.

Pan and amplitude envelopes work on the usual "rubber band" principle (although a "smooth" function can round off any corners), and, amazingly, you can use one

continued on page 130

Introducing WaveCenter/PCI

lour

OK, you see what's happening: digital mixers are looking pretty cool. After all, they've got incredible sonics, built-in effects, and the automation capabilities you could only dream about before. But if you hook that puppy up to the NoiseRacket analog soundcard that came with your computer, you're right back in ****ville. (Rhymes with "Snapville.")

Here's a better idea: keep it digital with our new **WaveCenter/PCI** card. It has all the connections you need to integrate your digital mixer into your computer-based studio. Transfer up to 10 channels of digital audio simultaneously using ADAT lightpipe and S/PDIF, with 24-bit resolution on all channels. Use one of the built-in MIDI ports for mixer automation, and the other to connect your synthesizers.

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5

CIRCLE 98 ON FREE INFO CARD



ARBORETUM SYSTEMS RAYGUN NOISE REDUC-TION SOFTWARE

while a Reset button is used to clear the meters' clip indicators.

Basically, all you need to do to clean up a soundfile is adjust the threshold, attenuation. and/or sensitivity settings from their initial zero settings, while

continued on page 132

BY DAVID MILES HUBER

One of the coolest and most easy-touse noise reduction apps to come across my computer screen in a long time is Ray Gun from Arboretum Systems. Ray Gun can be used to reduce such nasty offenders as noise, clicks/pops, and hum --- all in real time using simple, on-screen controls. In addition to all this, I really have to

provides an easy way to remove from your soundfiles

tip my hat to the folks at Arboretum for offering Arboretum Ray Gun on a single Mac/PC disc in multiple platform and plug-in flavors (including DirectX, AudioSuite, VST, and Premiere plug-in formats). For those want to use Ray Unwanted Gun as a stand-alone program, an executable ver-**NOISE** sion is also included that runs on either a Pentium or Power Mac

> computer. On the fileformat/samplerate side, Ray

Gun supports 16-bit audio files in WAV, AIFF, and SDH formats at sample rates of either 44.1 or 48 kHz.

THE CONTROLS

Ray Gun is made up of four modules (Noise Reduction, Pop, Filter, and Output) that can work separately or in any combination for use as an all-in-one audio restoration toolbox.

Noise Reduction: Reduces broadband noises (such as tape hiss, air conditioning and fan noise, and other background sounds)

using a downward expansion technique that's derived from Arboretum's high-end Ionizer noisereduction system.

Pop: Designed to clean up scratchy vinyl LPs, remove static crackles, fix clicks, and repair other transient noises by using a transient detection algorithm.

Filter: Used to remove low-end rumble, 50cycle (European) and 60cycle (North American) AC hum, these filters perform simple notch filtering using a setting that's not user-adjustable.

Output: Lets you monitor and control the final sound level. A Lock button can be used to group together the stereo faders,



MANUFACTURER: Arboretum Systems Inc., 75 Aura Vista, Pacifica, CA 94044. Tel: 650-738-4750. Web: orboretum.com.

APPLICATION: Noise reduction software for Mac and Windows 95/98/NT.

SUMMARY: A plug-in or stand-alone program that can remove noise, pops, and hum from a soundfile in real time.

STRENGTHS: Simple, straightforward, and effective to use. Sonically, it has fewer sideband artifacts and a fuller, more natural sound than many noise-reduction packages I've heard. The de-popper section is also really good. A personal thanks for putting Microsoft's DirectX Ver. 6 on the disc.

WEAKNESSES: The Filter module could be more effective, as it had little effect in reducing hums and buzzes.

SYSTEM REQUIREMENTS: Power Mac running at 120 MHz or faster with OS 7.6 or higher and 16 MB of free RAM, or a 133 MHz or faster Pentium computer with Windows 95/98 or NT 4.0 and 4 MB of free RAM.

PRICE: \$99

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World Radio History

AM62 Tube Multipattern Condenser Mic

EQ

CAKEVVALK **PRO AUDIO 9**

Don't let the slick 3D graphics of Cakewalk Pro Audio 9 fool you into thinking this update is just a makeover. Low-level code has been developed that dramatically improves the program's audio capabilities. On top of that, improvements to a wide range of core features make producing audio and MIDI tracks easier and faster than ever. For the feature hungry, Cakewalk has rolled the capabilities of its Guitar Studio software into Pro Audio to make this an all-inclusive upgrade for any of their users.

There's a wide variety of Cakewalk Pro Audio packages to choose from. From the very full-featured base program (Pro Audio 9) to the comprehensive Pro Audio

Suite (Pro Audio 9, Nemesvs Cakewalk's GigaSampler, LE, Audio FX 1, Audio FX 2, Audio FX 3, and latest Musicians Toolbox III), Cakewalk has production tools to update suit the MIDI composer, guiprovides a tar player, or audio engineer. The base program includes powerful features for Web-based media and audio-for-video (using sequencing AVI files). There's even the optional StudioMix hardware program controller for those who hate the mouse but love the flexiand a bility of software-based music production. Visit the Cakewhole lot walk Web site (www.cakewalk.com) and compare all more the features in detail before

making a final choice.

I tested the program on a 400 MHz Pentium II with 128 MB of RAM and Yamaha DSP Factory and SoundBlaster 64AWE Gold interfaces. I've been building projects in version 9 for a while, and upgrading the software from the Cakewalk Web site (version 9.02 is current at the time of writing) to keep on top of the changes, however minor. Thankfully, this was not a program that needed patches before you could trust your projects to it. If only my word processor was as reliable,

BY WADE MCGREGOR this review would have been written already....

Simple, but highly appreciated additions, such as hard disk and CPU usage meters, help

you push the performance envelope of your computing hardware and troubleshoot glitches. (If you think your computer is fast, just try getting 100 audio tracks off the disk at the same time — Cakewalk will support up to 256 tracks, if you've got the hardware!) However, the use of code (called WavePipe) to improve access to the sound card makes these meters far less busy than they would have been in previous versions. WavePipe is Cakewalk's term for the software that interacts with your Windows sound card at a very low (kernel) level to overcome the sluggish performance of Windows when trying to stream a lot of audio through hardware audio interfaces. The implementation provides an order of magnitude improvement in the responsiveness of the on-screen controls over previous versions of the program.



FIGURE 1: As in previous versions, Cakewalk Pro Audio 9 saves all of your screen window positions with the work file. The Patch track-parameter can now be used to prevent the audio stretching with MIDI tempo changes. Note the Zoom pop-up in the Audio View window, typical of enhancements to the mouse controls within the program.



FIGURE 2: The Console view includes access to plug-in effects, record metering, and practical sliders for effects sends.

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We loved Recording magazine's SampleTrak review so much, we had to rip it off.

BPM/VALU

EXII

PLAYLIST

AUTO SYNC

"Effects quality here is exactly what's needed for remix and dance music production. Filters and the ring modulator have a very "analog-ish" sound, and the time-based effects are clean and crisp."

SPECIAL

"A lot of attention has been paid both to sonic details and to real-time effects control. Case in point: not only does the scratch effect sound very record-like, but rocking the Edit I wheel makes it behave that way as well."

"The ST-224's internal noise is virtually nonexistent."

BANK 1

RANK 2

RANK 2

"Resample allows you to take material already in the ST-224, route it through the machine's effects and alter it, then save the results to another pad without routing anything out of the box...This feature takes the ST-224 and puts it into a higher league."

SOURCE

DAC Crowell Recording, September 1999 "In the end, the Zoom SampleTrak ST-224 is less like a phrase sampler and more like a little shrunken-down sampling workstation ...one that doesn't cost all that much more than those little loopers. Lots of KA-BOOM! for the buck."



LOOP/MARK



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"You wouldn't think condenser mics could get any more alfordable, but Marshall Electronics have dared to step into an already crowded bull pen with a deal you can't pass up."

Eddie Ciletti - Contributing Editor EQ Magazine, March 2000

"The MXE2003 strikes me as a mic that could very easily find a home in broadcast, ADR and Toley applications, in addition to a multitude of music chores"

Roger Maycock - Technical Consultant Mix Magazine, March 2000



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FIGURE 3: The AudioX Console view for the Yamaha DSP Factory. Note the hardware functions are a double-click away, but save screen space.



MANUFACTURER: Cakewalk, 5 Cambridge Center, Cambridge, MA 02142. Tel: 617+441-7870. Web:

APPLICATIONS: Windows 95/98/NT software for the production of music; interchangeably using MIDI or audio in project studio and postproduction environments.

SUMMARY: AudioX and WavePipe bring a whole new level of audio performance and control to this powerhouse sequencing program.

STRENGTHS: AudioX view offers hip graphics that don't get in the way of your mouse control. Optional StudioMix hardware does away with the mouse for the most common controls.

WEAKNESSES: Quirks to routing audio with DSP Factory hardware.

PRICE: \$429, upgrade pricing available. Optional StudioMix hardware \$299 as an upgrade to Cakewalk Pro Audio 9.

EQ FREE LIT. #: 105

The optional StudioMix hardware offers a great little dedicated interface for mixing and monitoring the recording. Its solid, wedge-shaped metal case contains nine motorized faders, which provide a direct method for mixing either audio or MIDI tracks. Connected to the MIDI interface of your computer sound card, the StudioMix is the most cost-effective moving fader package you can buy. Each fader is accompanied by two knobs (shaft-encoders) and a pushbutton that can be assigned to a variety of tasks in a little pop-up window from the Pro Audio menu or within the Console view. There are also a series of five broadly assignable

buttons that can access anything from punch-in/out marking to calling up a new mixer configuration. A rotary control allows fine control over the cursor position and is surrounded by a jog/shuttle ring that speeds you from one section of the song to another. If I pressed very hard while spinning the rotary control with my fingertip, I found it to be a little sticky, but I solved that with a little Teflontape applied underneath the center knob. Above these controls are dedicated transport controls that are switchable between controlling Cakewalk and an MMC-based external recorder. On the top right of Stu-

dioMix, four knobs give you a studio interface for a simple stereo sound card. These provide level control over a stereo line input (RCA), a mono microphone input (XLR), sends to and from the stereo line input of your sound card and your mastering recorder (RCA), and a monitor level control (RCA for stereo output to a monitor amp and a 3.5 mm mini-jack headphone output), which also switches between the send to the card and playback from the master recorder. This little box not only makes a stereo sound card into a practical audio production tool, it brings that analog console feel to your

continued on page 128

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Accelerate your Senses.

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World Radio History

PROGLIDE

24K gold contacts

UMX is the New Standard.

two stainless steel rails





DENON DN-2600F **DUAL CD PLAYER**

BY DJ RUSS REIGN

Recently, my promoter decided to organize a "moving" party to bring the New York underground club scene on the road. This party would create a loyal following by using the same DJs, lighting company, and promotional team in every city in which it appeared. Appropriately, the party was named "Movement," and the first one was slated for Albany, New York, at Elda's Nightclub, a large and classy venue where I appear monthly. This seemed like a great opportunity to road test the new Denon DN-2600F dual CD player, as well as to get comments from renowned DJ/remixer/producer Anthony Martinez of Future Primitive and Dot Dot Dot Records, who would be joining me that evening.

The 2600F's control unit can be rackmounted or freestanding, which came in handy as racks were lacking in the DJ booth. The link wire running from the CD player's control unit is recessed, so the unit stands level when placed on a flat surface. Incidentally, the unit I tested was the guinea pig for Denon's "drop test" at the NAMM show. It was released from a height of one foot to fall on a hard surface over 2,000 times during the course of the show, so durability is definitely not a question.

FEATURE THIS

The 2600F has more features than a seven-series BMW. Denon has retained the basic design and layout of earlier models, as well as the true "instant start" capability that's crucial for beat mixing. Illuminated buttons and a large fluorescent display, with both text and symbol indicators, are informative and readable under low-light club conditions.

The flawless loop capability found in the earlier DN-2500 model has been upgraded in the DN-2600F to a double-loop function; you can set two different loop points during a track and switch between them seamlessly in the mix. The loop cues also function as independent "hot starts." Any time you press one of these buttons during live play of a track, playback returns immediately to the cued point.

Knowing how a song ends is mandatory for beat mixing, so Denon built in an

tory for beat mixing, so Den end-monitor function that allows reviewing the final portion of a song at the touch of a button — no fast forwarding, no spinning of jog dials, and no scrambling to get back to your cue point. The unit even allows you to set the number of seconds you'd like to monitor, then stores that value in memory.

Sick of changing a track's pitch (speed or BPM) and getting the "Mickey Mouse" effect be-

cause this alters the key of the singer's voice as well? Denon's automatic keyadjust function lets you alter a selection's tempo, yet the key of the music remains constant.

More than just a CD player, this device gives DJs a good deal of creative flexibility

World Radio History

CAD Equitek E-350 Condenser Microphone

An unique powering system leads to an extremely wide dynamic range on CAD's new mic

BY STEVE LA CERRA

The Equitek E-350 from CAD is a multipattern, large-diaphragm condenser microphone intended for studio and live sound applications. Built around CAD's Optema OS-110 capsule with a 1.1-inch inside diameter and a three-micron diaphragm, the E-350 may be operated in omnidirectional, cardioid, or bidirectional pickup patterns, CAD's engineers have designed the E-350 (as well as other models in the Equitek line) using high-speed opamps in the preamp as opposed to FETs, claiming that this approach avoids the inherent nonlinearities associated with FETs and provides high gain. Although the E-350 may be powered via standard 48-volt phantom power, these opamps may demand higher current than some phantom supplies can deliver. CAD's solution to this increased current demand is a pair of internal, NiMII rechargeable batteries. Phantom power charges the batteries and the capsule then draws its power from the batteries - creating a reserve for use when the capsule needs more current (high-SPL transients, for example). In practice, this process is transparent, increases the dynamic range of the microphone, and allows it to operate without phantom power for up to six hours. The mic may also be powered using two standard 9-volt alkaline hatteries. CAD ships the E-350 with the ZM-1G shock mount suspension in a rugged, foam-lined, plastic case.

On the front of the E-350's body is a set of switches for power on/off, polar pattern, low-frequency rolloff, and 20 dB pad. Due to the nature of its battery-supply, CAD recommends connecting the microphone to a phantom source for a period of 12 to 14 hours (with the power switch in the "off" position) in order to charge the batteries before initial use. An auto-detect circuit shuts the mic down in the absence of phantom power; auto-shutoff may be defeated by moving a small internal jumper that puts the mic in "manual" mode, enabling the power switch. CAD's brief but useful instruction sheet accompanying the E-350 makes it quite clear that if the mic is running on 9-volt alkaline batteries, phantom power should *not* be used.

As per CAD's instruction sheet, we charged the mics before use by connecting them overnight to our Yamaha 02R with phantom power switched on. Our first session with the E-350 was a live tracking session with bass, drums, acoustic guitar, and vocal for an upcoming CD from the group GodSalad. We used one E-350 set to cardioid on kick drum, placed outside the drum at the hole in the front head. No doubt about it the output from this mic was hot. Even with the 20 dB pad switched in, we barely had to crack open the mic trim on the 02R to get a healthy level to tape. Under such high-SPL conditions, many users will find that the E-350 can overload the front end of their mic pro, so the 20 dB pad is a welcome and necessary feature. The sound of the kick was wonderful using the E-350; it had a controlled, extended bottom end that never sounded flappy and mixed well with the bass guitar, while still capturing the "wrinkle" of the beater against the head.

Simultaneously during this session, we used another E-350 to record a reference vocal/acoustic guitar track from GodSalad singer Pete Schorr. Since we were using the one mic for both guitar and vocal, we set it to cardioid and placed it at chest height roughly two feet in front of Pete. The timbre of the guitar was clean and crisp with a steely, jangly quality to the strings that cut through the mix without sounding harsh. Since proximity effect was not noticeable at this distance, the guitar was a bit on the lean side of the acoustic spectrum. Schorr's voice on this same rough track was well balanced and smooth in the lower-

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mids. Sonically speaking, we could have used this guitar/voice track for the finished recording. In the end, we decided against it, since Pete's performance was a bit subdued.

Which brings us to the recording of Pete's lead vocal: He can be a total powerhouse at the mic, and his voice can sound shrill in the upper-mids when using the wrong mic. The E-350 was totally up to the challenge of capturing his voice, providing a smooth top-end with good presence and no shrillness. On this particular song we were worried that when he hit the second verse, every red light in the studio would turn on — but the CAD mic didn't give up and handled the SPL like a champ — even with the pad off. Since we could run the preamp gain low, the resulting recording was quiet. The E-350's low-frequency extension resulted in some stand-transmitted rumble that the shock mount couldn't quite absorb; switching the low-frequency filter cleaned up the rumble without



them by accessing the *EQ* Message Boards at

www.eqmag.com

changing the timbre of Pete's voice. When Pete got within about 3 inches of the mic, the lower-mids became subtly emphasized, but don't expect the E-350 to turn a tenor into a baritone!

Frequency response of the E-350 in bidirectional was consistent in the 0and 180-degree positions, making the rear of the mic sound more like the front than some other condenser mics. The omnidirectional pattern was uniform in the high frequencies, as demonstrated when using it on a percussionist with a very mobile shaker.

Two things annoyed us about the E-350's shock mount. First, the combination of the c-shaped ring securing the top of the mic plus the position of the XLR connector in the opening of the "c" restricts the mic's rotation to no more than about 20 degrees in either direction. This makes it (at times) a bit of a pain in the you-know-what to move into optimum position. If the mic body was smaller, this point would be moot, but since it's rather large, there can be minor challenges in positioning the diaphragm toward the source. Second, the elastic bands that hold the mic at the mid-section tend to let the mic slip down about 1/8-inch; just enough for the elastic to cover all of the switches. As a result, you won't be able to quickly glance at the mic and verify settings. At one point this became a problem because we thought the mic was set to cardioid when it was actually set to bidirectional.

Minor annoyances aside, CAD has come up with an all-around mic that sounds natural in a variety situations, is constructed to last, provides multiple patterns, and can be used in the field for ENG or sampling without requiring a separate power supply. CAD's implementation of the battery supply to provide increased current reserve to the preamp provides the E-350 with an extremely wide dynamic range. Add in the fact that its MSRP is a reasonable \$899. and the E-350 should definitely be on your audition list.

PRICE: \$899

For more information on the CAD Equitek E-350 microphone, or any of their other products, contact CAD, Inc., 341 Harbor Street, Conneaut, OH 44039. Tel: 440-593-1111. Web: www.cadmics.com. Circle EQ free lit. #107.



William Wittman is a multi-platinum Producer/Engineer, former Staff Producer/ A&R Vice President (RCA / BMG Records and Columbia / Sony Records), Musician and Songwriter. His career truly covers all the bases.

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Avalon Design VT-747SP Compressor/Equalizer

Avalon Design provides a high-quality mastering tool

BY MIKE SOKOL

Basically, the Avalon Design VT-747SP is a stereo mastering tool that combines a wonderful sounding optical element compressor using a class-A tube design with a basic 6-band passive filter equalizer with fixed but intelligently selected frequency points. Being a true stereo unit, there's no way to break it apart and use one side for mono vocals and the other side for bass guitar. But, if you really wanted to, you could pass single tracks out to it one at a time during a tracking or sub-mix session, depending on how you like to do such things.

Housed in a sleek-looking 2-RU package, the VT-747SP does have a lot of copper and iron in the construction, so make sure you have some help when you put it in your rack. And because it's a Class-A tube design, make sure you leave some ventilation above and below it. There are no noisy fans to disturb the quiet of your control room, but you do need to give it a little air to breathe. Operationally, the left side of the front panel is the compressor section with all the functions you could want in a professional unit.

EC LABREPORT

MANUFACTURER. Avalon Design, P.O. Box 5976, San Clemente, CA 92673. Tel: 949-492-2000. Fax: 949-492-4284. Web: www.avalondesign.com.

APPLICATION: Stereo compressor-equalizer for mastering

SUMMARY: Stereo class-A vacuum tube compressor with twin signal path (with tube bypass) and variable attack/release times and ratios, opto-compressor design, and stereo 6-band graphic equalizer utilizing discrete Class-A/passive design. Sidechain insert path with monitor/listen. Lots of muscle with a delicate touch.

STRENGTHS: This is one fine mastering tool in a box. You can change the signal path with the compressor pre- or post-EQ, and the analog gain meter is almost a work of art. Plus, the signal chain has a hard-wire bypass for any function you're not using at the moment. And it will accept signals up to plus 36 dBv, which makes for some serious headroom.

WEAKNESSES: None that I could find.

PRICE: \$2,495

EQ FREE LIT. #: 124

The TSP switch switches in the tube path if you wish, or it can be eliminated from the signal path. Controls for attack and release times as well as compression ratios allow you to tailor the action of the optical gain element, and there's a set of spectral controls for the sidechain, which can be used to zero in on a problem area such as vocal sibilance while leaving the bass untouched. And, of course, there's a make-up gain control. A large meter that indicates the amount of gain reduction taking place dominates the center of the panel.

To the right of the front panel is a straightforward 6-band equalizer

and a stereo output level meter. A main output control and bypass switch round out the functions. The back panel has XLR connectors for the I/O and a strap that can lift the power supply from the chassis ground, this being especially helpful in a rack with various grounding schemes.

Within minutes of unpacking the box, I decided it was time to light the fires (literally) and peruse the manual while all those little electrons got warmed up and began their speed-of-light journey from the cathode to the anode. (Just how fast are the electrons actually traveling in a tube anyway? I'm guessing it's at the



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106	Gemini Sound Inc.	19	732-969-9000	18-19	TASCAM/TEAC America, Inc.	58	323-726-0303
101	Grnndmn's Music & Sound	20	800-444-5252	53	TC Electronic	59	805-373-1828
70	Great River Electronics, Inc.	21	612-455-1846	33	Telex Communications	60	800-667-3968
125	HHB Communications Ltd.	22	310 319-1111	116	The John Hardy Company	61	847-864-8060
113	JBL Professional	75	818-894-8850	94	The Jahn Lennan Sang Writing Cantes	st XX	www.jlsc.com
37	JOEMEEK	23	877-563-6335	124	The Recarding Workshop	62	614-663-2544
2	Korg	24	800-335-0800	39	Yamaha Corporation of America	63	714-522-9000
93	Kurzweil	25	800-421-9846	25	Zzounds Music	64	708-442-3620

speed of light in a vacuum, but possibly the control grid bias and signal fluctuations change the speed as well as the density of the beam. But I digress ...)

Being in a hurry, I just pick a B-52's CD off the pile and cue up Private Idaho. But, while the equalizer section really worked well, the compressor function wasn't happening for me. A quick mental (and meter) check confirmed that the song already had been squeezed to within a decibel of it's life anyway, so it wasn't appropriate material to judge

compressor action. Back to the stack of un-mastered DATs, which produced several candidates for level control. I had a nice Celtic piece with a beautiful image and tone, but which was way too low in overall volume - even when the peaks were slapping zero. A little deft action on the compressor with some make-up gain and the song still had nice transient peaks, but the VU meters on my console went up by 10 dB. Very nice. Then I patched in the equalizer after the comp for a little final sheen on the top while



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taking out some obnoxious 5 kHz "presence" from the original mix. (Oops that was me mixing the tracks originally...what could I have been thinking?) A little bottom roll-off completed the sculpting, and it sounded far better coming out of the box then it had going in. Remember,] was only doing 1 to 2 dB changes — a finesse fix rather than a sledgehammer fix.

Now on to some "metal" tracks that I was supposed to "sledgehammer" into shape. Basically, the artists wanted the tracks to be as loud as (in)humanly possible, but still fit within the 16 bits of a CD. Oh yes, the bottom in the mix was a little thin as well (don't monitor on speakers with enhanced bass, or you'll get thin mixes). I left the tubes in the signal chain in the path, added a few tweaks of the graphic equalizer to put the bottom up where it needed to be, and folded in a little extra sizzle on the top. Since I was able to reposition the compressor after the EQ section, I could go for massive-aggressive compression without peaking that sounded simply huge. Yes, it was a sledgehammer fix — but you do what you gotta do.

To recap, like an iron fist in a velvet glove, this box is capable of great gentleness most of the time, but can be called upon to deliver maximum "iron" when needed. I personally don't like digital compressors, having gotten used to some great classic analog boxes over the last few decades. (I collect what used to be called junk, but is now considered classic gear.) Call me retro, but I still like to twist the knob, watch the meters, and hear the action. I can't get used to setting compressor functions using a mouse, and even digital compressors with real knobs seem to run out of steam before I can get the job done sometimes. Computer plug-ins and wannabe processors can be the flavor of the month, but you could buy a VT-747SP and count on it for 24/7 operation for the next few decades. And I don't think you would ever get tired of it. Highly recommended.

Mike Sokol is a musician/engineer/wordsmith who still has the Hammond B-3 and Leslie from his youth (you may be appropriately envious). He's looking for an Ampeg SVT head to complete/restore his collection of retro gear. Let us know if you've got one collecting dust ...

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Microboards StartREC 400 **CD-R Duplicator**

Turn your project studio into a CD duplication house

BY MIKE SOKOL

Duplicators are typically rather uninteresting. Designed to be the digital equivalent of an office paper copier (dare we say Xerox machine?), most of them have no more functionality than allowing you to put a copy of a master document in the input tray (which sometimes was a face or other body part) and making as many clones as you needed.

The StartREC 400 goes a good bit beyond the basic copier functions because it not only does the "clone" thing quite well, it also has a quite extensive I/O section. This allows you to do a trick like dump raw tracks from a DAT recorder directly into its hard drive, trim the tracks to proper length, place them in the desired order, and make an internal "master" for later duplication.

The front of the rack-mountable unit has room for six drive bays. The "input" unit is a 40X UltraPlex CD reader, which is mounted directly across from a drive-bay-sized control panel. Optionally, ordering this product with a 32X Plexor Reader drive will allow it to copy CD+G karoake discs if required. Mounted in the bottom drive bays are four CD-R recorders rated for 8X recording. The back panel will accept analog signals in both consumer (RCA –10 dBv) and pro (XLR +4 dBu) formats. Also included are ports for digital signals in S/PDIF (optical and coax) and AES/EBU (XLR).

Initial operation is very simple. If you already have a CD or CD-R master that you want to clone, you put it in the CD reader and select the Copy menu, which prompts you for the last copy function you setup (in this case CD to CD). Then you pop in up to four blank CD-R media blanks, and hit the Copy button once again. A few minutes later, the recorder trays will open, exposing the four freshly minted CD-Rs.

If, for some reason, there's a misfire in one of the recorders (as in when I put in a CD-R upside down), then that particular drive drawer will remain closed, notifying you of your transgression (or a bad piece of media). There are also a number of other menu options available on the two-line display. For instance, you can select 2X, 4X, or 8X recording, or choose a master previously stored to the internal hard drive. In that case you select HD to CD recording from the Copy menu and proceed as before. In terms of getting around in the menus, I needed to keep the reference chart handy, as I don't like stepping though multi-level menus. But if you worked with this box on a day-today basis, it would become pretty intuitive, since all functions are only a few buttons away.

If you need to do some simple track editing, then the StartREC has you covered. For instance, while I normally dump all my raw DAT tracks into a workstation for track selection, it's possible to put a full DAT tape worth of tracks into the StartREC's internal drive. Then you can pick the start and stop points and insert CD program numbers as needed. This can be a little confusing on a two-line display with a couple of menu buttons, but I've been spoiled by CD mastering software such as Sonic



Foundry's CD Architect (where you can visually see and control virtually every aspect of a CD master).

Nonetheless, I did a full project just on the StartREC 400 from DAT to CDs without taking it through my workstation first. I selected the input port as the AES/EBU connector and fired up an HHB Portadat with an hour-long performance tape in it. Within two hours. I had selected and input each of the tracks and selected the start/stop times. While I couldn't get the single-bit editing accuracy achievable from a good WAV-editing program, I made a perfectly acceptable master CD-R of the piano performance. Then we just loaded in a few blank discs, selected HD to CD record, and pressed the Copy button. We were done and the client was happy.

In all, I put a few hundred discs through this box, and it never failed to deliver perfect copies as long as I put the CD-Rs in right-side up. I did try a variety of different brand media, which varied from "Branded" Sony, BASF, and Imation discs, all the way down to no-name

LABREPORT

MANUFACTURER: Microboards, 1721 Lake Drive West, Chanhassen, MN 55317. Tel: 612-556-1600. Fax: 612-556-1620. Web: www.microboards.com.

APPLICATION: CD-R recorder/duplicator.

SUMMARY: A one by four 8X duplicator with internal hard drive and basic editing functions.

STRENGTHS: Not only will it make four duplicates from a CD master or the internal hard drive, but it will accept professional and consumer analog, S/PDIF (coax and optical) and AES/EBU inputs, and record the tracks to the internal hard drive for later track editing and duplication. Overall basic operation is as simple as pop-'n'-go!

WEAKNESSES: Won't write CD-RW media, only CD-R. Editing track information is a little confusing on a two-line display.

PRICE: \$3,995

EQ FREE LIT. #: 125

blue/silver generic discs. Microboards officially recommends Taiyo Yuden CD-Rs as their media brand of choice.

Again, if you need some simple track editing, this box will do the job. But if you need serious mastering functions, then you're better off loading your tracks into a workstation and making a master disc. After that, it's simple enough that my 6-year old twin boys could operate it (please, no calls about child labor laws).

Nicely executed, and it should give years of dependable service.





But Who is Paying Attention?



PHOTO BY STEVE JENNINGS

There are plenty of places to post your music on the Web, but is it worth the effort?

BY JON LUINI AND ALLEN WHITMAN

This wet winter month we mostly look at e-zines, some of which encourage musicians to send in their own work for editorial review. The sites below do this in varying ways with varying degrees of success. Though it's always worth your time to submit music, don't expect too much. Who is paying attention? As our FezDad always says: "Most of life is just showing up." That's a good argument for playing live. But hey! This ain't Gig, this is a project studio magazine. It's for all you twisted studio rats, holed up in a dim room with no ventilation, putting the finishing touches on your sonic masterpiece. Get the hell out of your room! Take your music to the people! Vibrate the atmosphere with modulated pressure waves! Get used to looking like an ass! It's good for you.

> First, an upload site named Vitaminic... Vitaminic's tagline is: "Thousands

Of Bands In Over 150 Genres (The Music Evolution)." Yup, that's right. They got all 150 genres. They're all here. There's got to be a better way to categorize music. Ask yourself how the brain would do it. Imagine wanting to hear some music at home. You go to your CD rack and scan the titles. If you alphabetize your collection, this exercise is less effective because you already know where everything is. Depending on how you're feeling at that exact moment, a certain kind of music is going to appeal to you. If only there was a way to categorize Web-centric audio by emulating the way the brain works. Anyway, back to Vitaminic. This Italian-based MP3 download site looks like most big music sites: too busy. There's news (more useless "value-added" content), free offers of physical promo CDs to mail you, information about the MP3 format, and actual downloads. The music downloads are accompanied by brief typo-laden explanations. It appears that English may not be the first language. Not that that's a damning flaw or anything

First we register. Of course, a signature is required on a 911-word legal document. We sign without reading. We're silly. It could've said anything. As you read this we could be dressed in jodhpurs and Nehru, wielding pickaxes by the side of a two-lane blacktop in northwestern Alabama. Just a thought. Then we fill out the Vitaminic registration forms and lie about the phone number. (Ho-hum, why do sites insist?) When all is done, Vitaminic sends an email with our login name and password, as well as a tech support contact whose e-mail describes them as our "band manager." If they only knew what that really means! The process up to now has been very fast. We log on as a member and start to create our little area. The info supplied in the registration process is helpfully repurposed into these artist pages. A time saver! Uploading is straightforward, though there is no alert message on how long the process is taking. Finally, the music and the site are up, and all that remains is the much-ballyhooed stampede that, lifting us like a huge wave, will carry us to rock stardom.

Bottom Line: Vitaminic has ripped

most of its ideas from the book of riffage.com and MP3.com. It's quick and simple, but suffers from the same malaise of those other sites: Who is listening? And how can you, the artist/musician find fans? You've seen this site before and it's nothing new. But, hell, if you've got the time, why not upload? The more the merrier! Visit <u>www.</u> **yitaminic.co.uk**.

Okay, let's check out some e-zines.

Zine: Ink Blot Magazine URL: www.inkblotmagazine.com

Tagline: "Deep coverage of great music" Coverage: The site design is a little confusing, but the writing is good and there are many interesting MP3s. *Ink Blot Magazine* allows and encourages link trading and features several pages of eclectic band links with short explanations. The "Cream of the Crop" section features one band a month selected from submitted, independent music. Depending on how much music is submitted, your chances of notice are slim at best. This e-zine is definitely a labor of love.

Zine: Red Button

URL: www.redbutton.com

Tagline: "Independent e-zine and music channel"

Coverage: The downloads are all in the Liquid Audio format and the occasional text reads like record company promo 101. *Red Button* considers themselves an "artist management company," as well. A conflict of interest? Submissions are encouraged in the form of physical product or a link to the encoded song on your server.

Quote: "This site is free to all artists we want to work with. We only put up what we like and what we think has the greatest potential for success." *Red Button* is a trawler, gill-netting for big-pig halibut.

Zine: Snackcake!

URL: www.snackcake.com

Tagline: "The magazine devoted to music and snacks and those who love them"

Coverage: Cool indie site, loose and charming, if rather San Franciscocentric. The editor also works for corporate rockzine *SonicNet*. The writing is honest though somewhat disjointed. Many interesting independent underground bands are highlighted. There's no offer for submissions of music and no downloads are available. There's an eclectic link page that's worth the whole site. The <u>hamsterdance.com</u> site is recommended!

Zine: In Music We Trust URL: <u>www.inmusicwetrust.com</u> Tagline: "In music we trust"

Coverage: The writing is sophomoric and redundant but the editors of IMWT genuinely love music and display their passion by writing glowingly about their favorite artists, a pretty eclectic mix. There is a search engine to guide you through the 2270(!) articles on the site. Banner ads from unrelated sites (jewelry sites and forbes.com, etc.) point to the revenue model. Almost all the artists are independent. Since there is no offer for music submissions and no downloads available, we're curious to know how they find these bands. But, more importantly and somewhat rhetorically, why should we care? Who has time to go through all this stuff? The "subscribe page" asks for a snail mail address to send e-mails. IMWT posts a printed message declaring their solemn promise not to give away user data. So what's the point of requesting the snail mail address? The message board is either faux-fan emails posted by interns at major labels and other Web sites or straight promotional boilerplate. The 49 messages posted since November of 1998 are displayed in a very confusing layout. Is anyone listening?

Overall: Passionate and ineffectual editorial style, unfocused design. They seem to be saying: "Don't call us, we'll call you."

Other relevant places to submit music to be reviewed:

Site: www.listen.com

Tagline: "The music download directory" Coverage: Everything. This site is huge. They take recommendations/submissions for review. Upon review, the artist is entered into the listen.com search engine and the site links to downloads. Good writing, exhaustive listings, you could easily get lost in here. The lists of bands and downloads seem endless — and that feeling describes perfectly what's wrong with the Web. There's too much information and no intelligent, rational, and synchronistic sorting method along the lines of the human mind. No Dewey Decimal system here, in the end the Web offers a mountain of music, impossible to climb. That's good, right?

Site: www.cdreviews.com

Tagline: "Internet music review service" Coverage: Very straightforward, simple, and concise with good solid writing. The site invites you to submit your CD and they will post reviews. Unknowns, cult figures, sidemen projects, and musicians dumped by labels (usually a very good thing for the musician!) pepper the site.

Feel free to address any questions, comments, rants, and tips about Internet audio to the FezGuys Threaded Discussion Area, <u>www.fezguys.com/tda</u>. We'll be there!



www.tannov.com

KRK FIREWIRE

continued from page 38

used to link six Di Series monitors to a laptop computer, and — when referenced to the position of a calibration microphone — diagnostic data can be collected about whether each speaker is positioned at the correct distance from the mic, operating across the correct frequency range, or wired with proper phase.

Essentially, KRK says the Di monitors are "future-proof" because they are online upgradeable. It's even possible that subtle modifications to the frequency response of a system could be made at the factory by KRK and then downloaded via Internet directly to the Di Series flash memory. Such re-tuning ability could even be used to tweak the speaker's response after initial break-in.

KRK will begin shipping the E7Di and E8Di to selected dealers by June 2000. Suggested retail prices were not available at press time. Owners of current Exposé E7 and E8 monitors may return their monitors to KRK for upgrade to Di Series technology with an expected turnaround time of 48 hours. For more information contact KRK at (tel.) 714-373-4600, (fax) 714-373-0421, or on the Web at www.krksys.com. Circle EQ free lit. #108.

ANATOMY...REMIX

continued from page 46

Once again, I was reminded that music is about collaboration, having fun, trusting your intuition, and not falling in love with what you do so you can be brutal during the editing process. Maybe I can talk them into doing some more remixes...

You can hear the remix "new york city 01" on the Battery Park Cologne 3.0 compilation, Harvest Records #7243 5 22885 2 6 [CD] or #7243 8 87840 6 1 [vinyl]. The "Dogs" remix appeared on a limited edition double-CD for Battery Park Festival attendees, but will also be included on the *Sexy World* CD. To contact B.W.A. for remixes, send an e-mail to remickz@syncom-productionz.de.

Craig Anderton has played on, produced, or mixed 17 major-label recordings, presented lectures on technology and the arts in 37 states and ten countries, and written 15 books. He is also the online content editor for the MPN Network (www.MusicPlayers Network.com).

GREGG MANN

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Superherorockstar record is going to be very big in Europe. I did a compilation album of top French hip-hop artists with DJ Mars, which is coming out this spring. Is there anyone, living or dead, that you wish you could record? Dead? Jimi Hendrix or Miles Davis. Living? The Artist.

What music would you like played at your funeral?

"Voodoo Chile" by Jimi Hendrix and "End Of The Road" by Boyz II Men. Do you have any advice for those entering this wonderful world of recording? Never believe that you can't get it done. If someone says that you can't do it, immediately believe that you can. Also, I found that having my own little mixer/recorder setup at home was very helpful. When I started out, I only had four tracks, but it forced me to balance everything on those four tracks. When I got to the big boards and lots of tracks, I had a much better foundation. And

one more thing: Learn how to record real, live instruments. What would you like Santa to bring you

this year? A really creative new artist — so we can break some new ground.

JERRY FINN

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them, so that's what they want to hear. How loud do you listen?

Extremely quiet. Like at conversation volume — probably 85 dB or so at the loudest.

How many mixes [of one song] do you do? If it's up to me, I'll do the main mix, a vocal up 1 dB, a TV mix, and an instrumental. I've worked with some producers that want to avoid conflict, so they'll sit there and print mixes all day to please every guy in the band, but all you're doing is prolonging the argument. You end up with a nightmare at mastering as you edit between mixes, so I try to really just get it right the first time. The instrumental comes in handy sometimes for editing out cuss words and things like that.

What studios do you like to work in? Conway (in Hollywood) is definitely my favorite studio. Before I went independent, I was an assistant there for about four or five months. When I finally went independent, I was so scared because I had only done the Green Day record. Still, it was just blowing up so huge and I was getting so many calls that I had to pursue it. Being realistic about the music business, I thought I'd have a red-hot career for six months and then be back assisting. When I left, I made them promise that when my career fell apart they'd hire me back as an assistant. [Laughs.] I still joke with Charlene, the studio manager, about that whenever I see her. "Are you still gonna hire me back when my career falls apart?"

TONY FERGUSON

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the recording; it's the music. And then, if the recording is good, it'll only enhance the music.

On a business level, how do new audio pros get noticed by a record company? They come with a studio or an artist. Like the way we found a very talented producer/engineer, Eric Valentine. His involvement in the Smash Mouth record that came to us knocked us out. The guy's one of the best up-and-coming producers in the business. So, again, it's the music that comes first.

Do record companies look to a talented individual as a "go to?"

Absolutely. Particularly if they do consistently well, and in this business a couple of records is consistent. It's not necessarily a decade of making hit records. A couple is enough to get you on the map and get you on peoples' hit lists.

How does one stay visible?

Align with a good representative to keep the name out there and solicit work. Also interact directly with an artist so you can bring talent to a label. A lot of producer/engineers do that.

It's commonplace for new artists to get their deals that way. A producer shows up with something that sounds like a record, even if it's not ultimately the record. It mitigates the risk to the record company.

Absolutely. And it's becoming more common. The role of the A&R person is becoming less involved in the initial manufacturing or developing of young artists, and it's going out to the farmlands of producers and engineers and studios.

Michael James is a producer/mixer/guitarist whose credits include Hole, New Radicals, and A.J. Croce. **OFULL COMPASS**

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NEUMANN

ENGLISH BEAT

continued from page 85

try EQ'ing the DI against it, put it out of phase and slide it in again; it will actually take parts of the sound out. So I'll always have a bass amp as well as a DI.

So you're nulling out certain frequencies between the two; in effect you're equalizing by phase cancellation. Yes, phase against EQ.

Do you typically compress or limit while you're recording the bass?

We compress *everything* a bit going to tape. A lot of the equipment in my studio came from Abbey Road, including the TG console — a very, very famous console which has 44 built-in compressors, though it's got very simple EQ, so most of the work we have to do in getting a sound is mic placement and microphones themselves. The board does have plenty of bass controls: one that's just bass plus and minus, and then there's a very accurate multi-choice bass frequency. There's just one other EQ, which is presence. It goes from 800 Hz up to 10k, but very broad, so it's not really 10k, it's more 5 to 15k.

Our main recording machines are two 16-tracks, which also came from Abbey Road. They were originally 1-inch 8-track, but we put in 2-inch head blocks one is the old hard head block, which I much prefer the sound of. I run the multitracks at 15 ips because the sound is so much superior to 30, and I use Dolby A — I don't like the sound of SR. The backing tracks are recorded on one or two 16-track machines, depending on how many tracks we use. I then usually transfer into a 24-bit Pro Tools system to cut together the final take, and then put it back to analog for the overdubs. Sometimes, though, the backing track is complete and I just start working on it.

Usually, I copy the 16-track master so it can be put away and not played until the final mix. On the copy, I'll do stereo drums, mono bass, and stereo guitars and end up with however many tracks are left for overdubs. I'll overdub on that 16-track, get the masters of the guitars, the vocals, whatever, compile them in Pro Tools, and put them back to a fresh 16-track that also doesn't get played until the mix. The disadvantage of going in and out of Pro Tools is far outweighed by the advantage of the editing control you have over the individual sounds.

What's your mix format of choice?

I mix to 1/2-inch analog 80 percent of the time, but I mix to Pro Tools the other 20

percent of the time simply because it's multitrack; you put down your backing track, you put down your vocals in stereo, you put down your backing vocals in stereo. You can split it and end up with a submix master.

I know the mix is important, but, in an ideal world, it's just a formality. I always want the rough mixes we've done up to the mixing stage to sound pretty good. Sometimes on the way to Pro Tools or when putting it back to tape, I've put signal through the desk and re-EQ'd it, so I've ended up with it at least partly submixed. So you should be able to throw up all the faders and in five minutes it should sound good.

Presumably you're hearing a fair number of tapes that are coming out of project studios. What are the common mistakes that people are making in their recordings?

I don't think they're making mistakes at all. A lot of the project studio stuff I'm hearing is fantastic. It's been harder and harder for me now because I hear a demo that somebody's done and I think, well, what would I do to this? Okay, I can take it into the studio and re-record it, but I would like to keep the vocals they've done because they're really good, and I really like that guitar effect, let's try and keep that. It's not uncommon now to start with that demo and work on it.

The main disadvantage in demos is drum sounds. Project studios often don't have live rooms, they haven't got loads of mics. On some songs, that can work, but it's touch and go, really, because you're so limited by what you can do. You've got to record your drums on 12 or 14 tracks so you can sweep your overheads in for the second bridge and back out again.

But I think, in the future, most recording will be done at home; I really see it going that way. People will just go into studios to take something that 10 percent further that you can't possibly do in a home studio, where you've got 50 or 60 microphones to choose from, you've got large ambient spaces, you've got the big multitrack facilities where you can actually listen comfortably to 60 tracks at a time if you want to. I think more and more album tracks will be done at home, and then people will tend to polish up some of the tracks in the bigger studios.

This interview is excerpted from Howard Massey's new book Behind The Glass, soon to be available from Miller-Freeman Books.

ROCKET NETWORK

continued from page 36

it can be posted to the virtual studio, where other users can share it.

In addition to posting and receiving track data, users may also chat with each other simply by pressing a "Talk To" button; this initiates a private, one-on-one chat utility. Other conveniences include a Studios listing, which may be organized to show private and public studio categories, and a preview function showing additional information about a particular studio or user. A "Favorites" function provides a quick link to frequently visited studios and other users.

A Rocket Network Software Developer's Kit (SDK) is available for manufacturers who wish to include Rocket-Power in their applications, thus allowing their customers to access the Rocket Network. Among the tools provided in the SDK are the Rocket Network application programming interface, or API, a sample application, RocketControl 2.0, digital audio and MIDI "work" files for testing purposes, and debugging tools.

CONNECTING

Although a 56k or faster modem is recommended, a user can connect to the Rocket Network with any speed modem or device. Connections through popular ISPs such as AOL are supported. Internet Recording Studios support multiple levels of compression for transfer of audio data, and participants may choose the appropriate compression level based on their connection or needs.

Emagic and Steinberg have both launched their own-branded Studio Centers. Emagic is sponsoring the new Harmony Central Studio Center, which offers visitors a free download of Logic Rocket and access to public studios with the option to purchase their own private studio. Steinberg recently launched the Cubase.net (www.cubase.net). RocketControl 2.0 is currently available as a download, free of charge; RocketControl Professional is available for \$29.95; the Software Developer's Kit is available for interested parties through sales@ rocketnetwork.com. For more information, contact Rocket Network at www.rocketnetwork.com or 415-538-0123. Circle EQ free lit. #123.

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World Radio History

DANNY SABER

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important not to get too much sh*t too fast. That's why you've got a lot of producers who make one really cool record and that's it, it's over, because there's no real depth behind what they do. They just kind of get lucky or whatever, they sample some old record and put it out. There's no depth there, no experience to back it up. So the next time they get a situation, it doesn't work out — they're only capable of doing one thing because they don't have versatility. And versatility is something you get by experience you don't just wake up one day and you're versatile.

You've talked about experience a lot, but the fact of the matter is that most people are impatient and say, "I want to do it now - I don't want to wait to learn what I need to know."

Well, do it now! Nobody's stopping you. Do it! The more you do now, the more experienced you'll be later. That's the whole point I'm trying to make — do as much sh*t as you can. But be willing to put in the work that it takes to get good at something. Look at basketball players — you don't wake up one day and get in the NBA. But it takes a certain number of years for a basketball player to grow to be tall enough to be in the NBA. Even if they've got it all together in their head, they're still forced into waiting.

Well, that's why most guys don't make it. In music, things can happen faster. But the only people sh*t happens fast for are the people who are capable of writing some kind of good song or making a good record. Technical people like engineers don't really make it overnight. Engineering is not the arena you go into if you're impatient and you want to be rich tomorrow. Because, one, you're not going to get rich from engineering — very few do — and, two, you're not going to pick up one day and know it all. It takes time to absorb all these things — you can't just wake up one day and be great at it.

I talk about this all the time to people who see where I'm at and think I just woke up one day and I was here. It doesn't work that way. This was a passion for me and something I just unconsciously did, and I knew I would succeed. In many ways, the best thing that kids that are starting out have is the naïveté of not knowing what they're up against; that's their best ally. When I was a kid, the most positive thing I'd ever get was, "Yeah, you should really go for it and try it, but you need something else in case it doesn't work out, because a lot of people are trying to do what you're trying to do." In my head, I would think, "F*ck you, I'm going to make it, I'm not everybody else." *That's* what you need. That's what gets you through it.

Self-confidence is what you're talking about.

Yeah, exactly. And patience is important, too. That's something that I had to learn, because I was the same way - I wanted it all right away, and it didn't happen that way. But you've got to let things come to you, you can't force the issue on everything. The things you have control over, force the issue on. But the things you don't - like if the guy who heard your tape is going to call you back - let it f*cking go. Because, ultimately, he's not going to call you back if he's not going to call you back. And if he likes your sh*t, he is. You can't do anything about that. But what you can do is you can work on your sh*t and get better at what you do. That way, you give yourself the best possible chance to get in a situation where you're doing what you want to do.

This interview is excerpted from Howard Massey's new book Behind The Glass, soon to be available from Miller-Freeman Books.

CAKEWALK AUDIO 9

continued from page 104

computer-based mixing tasks. The MIDI control assignment is fixed for each control, but, combined with the integrated functions of Cakewalk Pro Audio, it's an amazingly useful control interface.

As in previous versions of the program, fader and button grouping is available with a right-mouse click on a control. This is especially important when using the StudioMix, as you can combine channels to create moving-fader subgroups that can even have custom properties for each channel within the group. This allows the ninth fader to become the master of one output or of every bus in your on-screen mixer. However, there's no communication of grouping between the Console View mixer and an AudioX mixer, nor does the StudioMix address the AudioX mixer.

What's an AudioX mixer? This is a new standard for extended control of more sophisticated audio hardware, such as the Digital Audio Labs CardDeluxe, Sonorus STUDI/O, and Yamaha DSP Factory interfaces. AudioX extends the program's access to these devices to include making use of the multichannel I/O patching and on-board DSP functions. This requires an AudioX driver from the hardware manufacturer, but steps well beyond the basic functions supported by a standard Windows driver. (Although I did still have to resort to the DSP Factory's patchbay to kill the S/PDIF input.) Cakewalk has taken a very active roll in pushing the computer industry to adopt more sophisticated (and unified) drivers for professional audio work (check out the white paper on their Web site).

The program now includes the features of the Cakewalk Guitar Studio software, such as a photo-realistic fretboard displaying MIDI notes, the Amp-Sim plug-in (a "lite" version from Cakewalk's Audio FX 2 guitar amp simulator), the NTONYX Style Enhancer plug-in (also a "lite" version), a tuner (not that guitarists ever need that), and even a de-glitching filter for those with the nerve to generate MIDI directly from a guitar. While the guitarist is tuning, you can quickly put together the drum tracks using a drag 'n' drop plug-in called Session Drummer. Just select a drum loop/style from a vast array and drop either MIDI data or audio into a track, even in real-time.

It takes 519 pages of the well-written manual to cover every feature of this powerful sequencer, so I won't try to list them all here. However, the combination of the manual and practical on-screen help gives the new user and veteran alike access to enough information to make use of even the more esoteric features. As always, Cakewalk extends the program's functionality to extremes with the CAL programming language for those who want to build their own shortcuts. For the rest of us, a quick reference card is supplied for keeping the comprehensive set of keyboard shortcuts in view.

Cakewalk hasn't ignored the need for a session to end up on the Web, with a mixdown mode for MPEG (MP3), RealMedia (RM), and Windows Streaming Media (WMA) file formats. If you need to save hard drive bandwidth, it's also easy to submix tracks and bounce them down.

If you're new to Cakewalk, you may want to spend some time in a dealer's demo room checking out the tutorials to get a feel for this software. If you already use a previous version, how could you resist the upgrade deals Cakewalk offers, especially the price of the StudioMix hardware? You might even throw away that mouse, once your fingers get used to pushing real buttons and faders again.

Wade McGregor is a principal consultant for Mc2 System Design Group, a sound system design firm based in North Vancouver, BC. For more info, visit their home page at <u>www.</u> <u>mcsquared.com</u>.

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STEINBERG WAVELAB

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track's data when "ducking" on another track. For example, a solo guitarist's voice on one track can lower the level of the guitar on another track only during vocals.

As for effects, individual clips can have up to ten VST-compatible effects (DirectX, and, oddly enough, WaveLab effects will not work, except globally through the master section). Effects can either insert (*i.e.*, the clip runs through the effect), or work on a "send" basis where an envelope determines the wet/dry balance.

OTHER SUPPORT

WaveLab supports a wide range of CD recorders; rather than list them here, check for compatibility at www.vob.de/us/products/ consumer/WizardGold/devices.htm. Sampler support is also rich, including products from Akai, Ensoniq, Roland, EMU, Kurzweil, Yamaha, etc. For an upto-date listing, go to <u>http://</u> service.steinberg-na.com/knowl edge.nsf/show/wl sampler list

WAVELAB GROWS UP

With version 3.0, Steinberg has taken its digital audio editor to a new level. The program's overall feel is more solid than ever — and more comprehensive. For mastering, it has become my editor of choice (although Sound Forge still gets the nod for audio-for-video projects and some Web projects, thanks to its AVI and multiple file format support).

In the current world of leap-frogging revisions, WaveLab has raised the stakes and set a new standard in what a digital audio editor should do. It's a solid, professional program that has evolved from being "another digital audio editor" to being in a class by itself.

SERATO REVIEW

continued from page 101

Pro Tools versions, selecting an odd number of tracks could cause a crash.

Pitch'n Time was a welcome addition to the post house I was working at, as it was immediately used to stretch a spot — without pitch change — from 13.5 seconds to the required length of 15 seconds. We simply typed the requested length into the Length Output section and processed it. The plug-in provides a useful alternative to the standard time-compression and expansion available to Pro Tools users, and it effectively handled any task I required it to do. Dramatic shifts, such as pitching up 200 percent from the original pitch, will create a phasey effect, but that's a huge pitch-shift. I found the effect could actually be used creatively.

For example, I took a drum loop and pitched it down an octave without changing its length. By applying some creative EQ to take out the hihat of the pitched sample and combining it with the original loop, I got a really huge sound. The same approach can be used on vocals: pitch-shift two tracks, one up a few cents and one down a few cents, to create a nice rich sound. Pitching the vocals down in octaves produced some great sound-design effects, and I got some really wild phasing effects on another loop by pitching it up an octave, then back down an octave, and mixing it with the original.

This is really a great plug-in for remixers and sound-designers, and its obvious applications for post and audio repair will surely make it a useful addition to anyone's Pro Tools arsenal.



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World Radio History

ACROSS THE BOARD

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over. Counting how many samples are at full level does this. If just one or two samples are at full level, the possibility exists that the signal just barely reached full level and then went back down again with absolutely nothing wrong.

If you have a full level signal for more that two samples, then the only possible explanation is that the signal was still going up, but there aren't enough bits to record it. The result is a chopped-off top (or bottom) to the waveform. This is 100 percent distortion. The reason it is 100 percent is because none of the original detail is there, only 100 percent artifacts. If you isolated this piece of the signal, you would only get a full-level DC component and no modulation.

Professional digital meters in highend hardware have settings for the number of samples allowed at full-scale before the over lights come on. The Sony DMU-30 was the standard by which all others were measured. All of the CD mastering facilities used this exact same meter for detecting overs. This is the same meter as the meter section built into a Sony 1630, which was the machine that produced the CD master that was sent to the CD manufacturing plant. The standard adopted was four samples at full level or more was unacceptable.

In the early days of CD manufacturing, the plant would not accept a CD master that contained overs. They would bounce the master and have the mastering facility make a new one. If early CD players were hit with an over, they would produce a loud pop or click. The severity of the pop varied depending on the CD player. Overs were illegal.

If the master sent to the mastering facility was full of overs, the mastering engineer could turn down the final output by 0.1 dB and, even though the flat tops were still there, the over lights would never come on because the level never used up all of the bits. When the rap records and hip-hop craze flooded the market, every artist wanted their CD to be louder than the competition. Limiters from 1994 were not fast enough to crunch the audio without making it sound like a bad FM station. But you could raise the level 6 dB or more and let the 16 bits fill up until they clipped the signal. Just like a peak limiter set to zero attack time with infinite compression ratio. The results were that the over lights came on and stayed on until the entire CD was played.

The mastering guys just brought the final level down 0.1 dB (now you can bring it down 0.01 dB) and the over lights would extinguish. Any digital device such as a digital console or digital EQ will allow you to lower the level by 0.1 or 0.2 dB. The TC Electronic Finalizer allows you to set the maximum "never exceed" output level in .01 dB increments.

Current CD players do not click or pop when presented with this square signal, and the chopping off of the waveform actually adds to the punch of the kick or snare beat. If the square-top waveforms are unacceptable to the artist, engineer, or mastering engineer, a highpass filter in the chain will remove the flat tops and smooth things out.

BACK TO COUNTING OVER SAMPLES

I won't go into binary counting here. For further information, you can search the Internet or cut off all but one of your fingers.

Approximately 6 dB before clipping, the most significant bit goes on. Some older meters turn on the over light when this condition is met. This means that the over light will warn you 6 dB before you clip the recorded signal. This seemed like a fair place to set the over lights for a consumer machine. Keep them out of trouble by showing the over lights early. The first Sony PCM-2500 DAT machine was this way, which was based on the first consumer machine. If you want to be closer to the actual over level, you have to look at more bits. If you look at two bits, you are about 3 dB away. If you look at 15 bits (the 16th bit is the sign bit telling the converters whether the sample is positive or negative), you can give an indication of when the bits are all used up. Pro Tools counts all of the bits before the meters show over.

It takes some additional hardware or software to actually count the number of samples that have reached the "over" status. The meter must have a memory of what the last bunch of samples was (or just whether they were over). If you have the meter set to light up the over light when six samples in a row are over, then the meter must remember the last six sample conditions. The Sony DMU-30 meter, the meters in the Apogee AD-8000, and the Mytek digital meter are some of the hardware meters that count over samples before lighting the over light.

So, now you understand why you can record something on a Fostex D-10 DAT machine with no overs and, when you play it back on a Sony PCM-2500, the over lights are on *all* of the time. The levels on the tape are no different, the sound will be no different, it's just those pesky over lights.

Most of the newer DAT machines have very accurate meters and even display digital headroom (how many dB away from over the loudest peak is). Sony and Fostex machines have this feature. If the headroom display says 3.1 dB, then you can turn your mix up 3.1 dB more before you run out of bits. On most DAT machines, the over lights do not lock on as they do on stand-alone converters and outboard digital meters. After you have played through a mix, you can look at the headroom display. If it says 0.0 dB, then you probably went over somewhere along the line. Play the mix again and watch the meters. I always make a pass to watch meters before I print the final mix.

The bottom line is that one or two overs here and there are not going to ruin your master. If you can't hear it, then it is probably okay. It will likely get fixed in mastering anyway.

That is it for this month. Roger, WILCO, over, and out!

RAY GUN REVIEW

continued from page 98

auditioning the results in real time; That's pretty much it. Once you're happy, simply apply the processing to your audio.

SET YOUR PHASER ON STUN, SCOTTY

Right outta the gate, I'm gonna say that Ray Gun rocks! The noise reduction section works some major clean-up magic, while producing little or none of the bird-chirping sideband artifacts that you often find caused by noise reduction apps and plug-ins.

The Pop section likewise obliterates most pops, however, it's much less effective at getting rid of the surface crackles that exist on older vinyl records (although the Noise Reduction section will often do the trick). The Filter section seems to be the weakest of them all. With the test files that I had to work with, I couldn't hear much difference when either the rumble or hum filters were switched in. This section might be more effective if Arboretum put in variable controls that could be adjusted by the user.

All in all, this simple, multi-platform plug-in is an absolute stunner. It's simple, it sounds great, and it's affordable. What more could you ask for?

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FEATURES-

- 18 simultaneous 24-bit ins and outs with support for 44.1 and 48 kHz sample rates
- 20Hz 22kHz freq, response ± 0.5 dB 2 channel. ALB nic/1/4 line inputs with *26 dB pad 48v phantom power, gain knob, and HP Filter at 60Hz
- 6 ch. line inputs (1/4") TRS balanced/ unbalanced w/ software controlled gain
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- 1/4-inch unbalanced line outputs channels 3-8
 Headphone output with independent gain control knob
 2 channel S/PDIF coaxial digital I/O
- 8 channel AOAT optical I/O can also be used as 2 channel optical S/PDIF

Pro Tools LE

- · Supports 24 tracks of 16 or 24-bit audio and 128 sequenced MIDI tracks Sample-accurate simultaneous editing of audio & MIDI
- Real-time digital mixing capabilities include recall of all mixing parameters, support for edit and mix groups and complete automation of all volume, panning
- mutes and plug-ins.
- Route and mix outboard gear in realtime
- MP3 and RealAudio G2 file support (Mac)
 Two plug-in platforms offer multiple options for effects



processing---- Real-Time AudioSuite (RTAS) is a hostbased architecture that allows an effect to change and be dynamically automated in realtime as the audio plays back. —AudioSuite is a file-based format, that

renders a new file with the processed sound. • Bundled RTAS plug-ins include, 1 and 4-band EQ; Oynamics II- compressor, limiter, gate and expander/gate Mod Delay - short, slap, medium, and long delays with modulation capabilities for chorus or flange effects and dither. AudioSuite plug-ins include Time Compression/Expansion, Pitch Shift, Normalize, Reverse.

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- Automatic sample rate conversion from 32 and 48kHz · Automatic CD Format Detection feature and user friendly interface provide one touch button operation Front panel trim pot and LCD display provide accurate
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- · Supplied 9GB internal drive allows 45 minutes of audio across all 24 tracks
- · Wide SCSI port on the back panel allows you to add multiple drives. A front 5-1/2° bay available for installing an additional drive, or an approved DVD-RAM
- drive for back-up. ViewNet MX: a Java-based software suite for Mac and
- PC offers DAW style editing of audio regions, dedicated system set-up screens that make set-up quicker and easier and track load screens that make virtual track management a snap. Connects to a computer via a standard Ethernet line
- · Can record to Mac (SDII) or PC (WAV) formatted drives, allowing later export to the computer. The Dpen TL format allows compatible software to recognize virtual tracks without have to load, reposition and trim each digital file.

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- >104dB Dynamic range
- 20Hz 20kHz frequency response ±.5dB
- .1 hr. 48 min. recording time on a single 120 tape
- · On-Board SMPTE synchro nizer - chase or generate timecode On-Board support for MIDI Machine Control

APOGEE Rosetta 24-bit A to D Converter

The high-end quality analog to digital solution for the project studio. With support for both professional and consumer digital formats you can e high-end quality analog to digital solution now record your audio at a higher resolution and with greater detail than standard converters found on MDM's, DAT's and DAW's. Ideal for mastering or tracking

FEATURES-

- 24-bit. 44.1-48, 88.2-96 kHz Sample Rate (±10%) 116dB dynamic range (unweighted)
 Improved UV22HR for 16 and 2D-bit A/D conversion
- FRONT PANEL:
- 96kHZ)selector 16-bit (UV22). 20-bit (UV22) and
- Power switch . Sample Rate (44.1, 48, 88.2, UCID

Transparent analog to digital conversion designed to bring your music to the next level. XLR balanced inputs feed true 24-bit converters for revealing all the detail of the analog source 16-bit masters can take advantage of the AD9624's noise shaping function which enhances clarity of low level signals

FEATURES-

24-bit precision A/D conversion • Support for 32. 44.1, 48, 88.2 & 96kHz sample rates • Wordclock sync input. Selectable 16-bit noise shaping .

B&H PAGE 3



- Editing · Built-in editing capabilities include cut, copy, paste, split and ripple or overwrite
- . 100 levels of undo

· Supports destructive loop recording and nondestructive loop recording which continuously records new takes without erasing the previous version Build-In Synchronization-

- TBUS protocol can sample accurately lock 32 machines
- together for 384 tracks at 96kHz, or 768 tracks at 48kHz
- · Can generate or chase SMPTE timecode or MIDI Time Code · Word Clock In, Dut, and Thru ports
- 1:0 Options-
- · Optional analog and digital cards all provide 24
- channels of I/D. There is one slot for analog and one for digital
- IF-TD24- T/DIF module
- IF-AD24- ADAT Lightpipe module IF-AE24- AES/EBU module
- IF- AN24- A-D, D-A I/D module with DB-25 connectors
- Software Updates-System updates are made available through a front panel Smart Card slot or via computer directly from the
- TASCAM web site. DA-78HR Modular Digital Multitrack



Code synchronization and a digital mixer with pan and level controls. A coaxial S/PDIF digital I/D allows pre-mixed digital bouncing within a single unit, or externally to another recorder or even a DAT or CD recorder. Ho to 16 DTRS machines can be synchronized together for simultaneous, sample accurate control of 128 tracks of digital audio

> · Internal digital mixer with level and pan for internal bouncing, or for quick mixes • Track slip from -200 to +7200 samples

- · Expandable up to 128 tracks (16 machines)
- Word Sync In/Dut/Thru
 Analog output on DB25 balanced or RCA unbalanced Digital output on TDIF or 2 channels of S PDI

24-bit resolution selector • S/PDIF-ADAT optical selector • Soft Limit on or off • 12-segment metering w/ over ondicator & Meter Clear switch . Level trim

REAR PANEL • XLR balanced inputs • 2 x AES/EBU for 88.2/96kHz 2 channel path, Coaxial S/PDIF, switchable S/PDIF or

ADAT optical outputs • Wordclock out

AD 9624 24-bit A to D Converter



outputs • 20-segment LED meters w/ neak hold & clip indicators • ALSO AVAILABLE: DA9624 24-bi D/A converter

Roland The all digital Roland V-Mixing System when fully

expanded, is capable of mixing up to 94 channels with 16 steren (32 mono) onbeard multi effects including COSM Speaker Modeling. Utilizing a separate-component design comprised of the VM-C7200 console and VM 7200 rackmount processor, allows the V-Mixing System to be configured to suit your needs. Navigation is made easy via a friendly user interface. EleyBus and EZ routing capabilities as well as a large informative LCD and ultra-fast short cut keys

- 94 channels of digital automated mixing (fully expanded)
 Up to 48 channels of ADAT/Tascam T-DIF digital audio I/D with optional expansion boards and interfaces Separate console/processor design
- Quiet motorized faders, transport controls, total recall of all parameters including input gain, onboard mixer
- dynamic automation and scene memory
- 24 fader groups, dual-channel delays, 4-band parametric channel EO + channel HPF

· FlexBus and virtual patchbay' for unparalleled routing flexibility

 VS8F-2 Effects Expansion Board -- Provides 2 stereo effects processors including CDSM Speaker Modeling. Up to 3 additional boards can be user-installed into the VM-7200 processor, for 8 stereo or 16 mono effects

VM-24E I/O Expansion Board -- Offers 3 B-Bus I/Os on a single board. Each R-Bus I/D provides 8-in/8-out 24-bit digital I/D, totalling 24 I/D per expansion board.



M Basic 72

- · Up to 16 stereo (or 32 mono) multi-effects processors using optional VS8F-2 Effects Expansion Boards (2 effects processors standard) sterer
- CDSM Speaker Modeling and Mic Simulation technology 5.1 Surround mixing capabilities
 EZ Routing allows mixer settings to be saved as templates
- Realtime Spectrum Analyzer checks room acoustics in conjunction with noise generator and oscillator
- Digital cables between processor and mixer can be up to 100 meters long- ideal for live sound reinforcement
- DIF-AT Interface Box for ADAT/Tascam -- Converts signals between R-Bus (VM-24E expansion board required) and ADAT/Tascam T-DIF. Handles 8-in/8-out digital audio, 1/3 rackmount size.
- VM-24C Cascade Kit -- Connects two VM-Series processor units. Using two VM-7200 processors cascaded and fully expanded with R-Bus I/O, 94 channels of audio processing are available
- exicon MPX-500 24-Bit Dual Channel Effects Processor

EFFECTS & PROCESSING

The MPX 500 is a true stereo 24-bit dual-channel processor and like the MPX100 is powered by Lexicon's proprietary Lexichip and offers dual-channel processing. However, the MPX 500 offers even greater control over effects parameters, has digital inputs and outputs as well as a large graphics display.

- · 240 presets with classic, true stereo reverb programs
- as well as Tremolo, Rotary, Chorus, Flange, Pitch, Detune, 5.5 second Delay and Echo Balanced analog and S/PDIF digital I/O
- · 4 dedicated front panel knobs allow adjustment of effect parameters. Easy Learn mode allows MIDI patching of front panel controls Tempo-controlled delays look to Tap or MIDI clock

Dual-Engine design

· 24 bit A/D-D A converters

24 bit A/D-D/A converters

· 24 bit internal processing

S/PDIF digital I/D, 44.1-48kHz

Balanced 1/4 Jacks - Dual I/D

t.c. electronic M-One Dual Effects Processor

The M-Dne allows two reverbs or other effects 20 incredible TC effects

to be run simultaneously, without compromising sound quality. The intuitive yet

sophisticated interface gives you instant control

Based on the Classic TC2290 Delay, the D-

patterns to be tapped in directly or quantized

Two is the first unit that allows rhythm

to a specific tempo and subdivision

ALL ITEMS ARE COMPLETE WITH ALL ACCESSORIES AS SUPPLIED BY MANUFACTURER CIRCLE 69 ON FREE INFO CARD

- of all vital parameters and allows you to create awesome effects programs quickly and easily
- Dynamics · Analog-style user interface

Multitap Rhythm Delay

Absolute Repeat Control

Up to 10 seconds of Delay

. 50 Factory/100 User presets

D-TWO Multitap Rhythm Delay

- including, Reverb Chorus Tremolo, Pitch, Delay and
- S/PDIF digital I/D, 44.1-48kHz
 Balanced 1 4' Jacks Dual I/D
- · 24 bit internal processing 100 Factory/100 User presets

VIDEO and PRO AUDIO

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C414 TLII

"Vintage TL"

AKG

FEATURES-

Combines the best of old and new: Clegendary C12 acoustics and the latest

generation of C414 transformerless FET electronics. Although similar in design

and shape to the C414BULS, the ILII features a capsule that is a faithful sonic

recreation of the one used in the classic

C12 tube mic combined with computer

aided manufacturing techniques that

from microphone to microphone

and figure 8 polar patterns

quality digital recording.

assure greater uniformity in response

· Cardioid, hypercardioid, omnidirectional

. Frequency response 10Hz to 20kHz

· Warm, smooth microphone that is suitable for high

C4000B

ELECTERET CONDENSER

his new mic from AKG is a multi polar This new mic from ANG is a multi-point pattern condenser micropone using a unique electret dual large diaphram

transducer. It is based on the AKG SolidTube deisgn, except that the tube

has been replaced by a transistorized impedance converter/ preamp. The

C4000B exceptional low frequency

Frequency response 20Hz to 20kHz

cludes H-100 shock

FEATURES-

transformerless output stage offers the

Electret Dual Large Diaphram Transducer (1st of its kind) • Cardioid, hypercardioid &

• Extremely low self-noise • Bass cut filter & Pad switches • Requires 12, 24 or 48 V phantom power

unt and wind/pop screer

PHOTO - VIDEO - PRO AUDIO

ANES MICROF

NT-2 **Condenser Mic** The RØDE NT2 is a large diaphragm

true condenser studio mic that features both cardioid and omnidirectional polar patterns. The NT-2 offers superb sonic detail with a vintage flavor for vocal and instrument miking. Like all RODE mics the NT-2 is hand-assembled in Australia and is available at a ugh price.

FEATURES-Dual pressure gradient transducer 1 capsule with gold-sputtered membranes Low noise,

transformerless circuitry • Dmni and cardioid polar patterns • 135dB Max SPL • High uass filter switch and -10dB pad switch . Gold plated output connector and internal head pins Shockmount, Flight Case, and Pop Filter included

 20Hz-20kHz frequency response A) audio-technica.

AT4047 **Cardioid Condenser**

The AT4047 is the latest 40 Series large diaphragm condenser mic from Audio Technica. It has the low self noise, wide dynamic range and high sound pressure level capacity demanded by recording studios and sound reinforcement professionals.

FEATURES-

Side address cardioid condenser microphone for professional recording and critical applications in broadcast and live sound

 Low self noise, wide dynamic range and high SPL
 Switchable 80Hz Hi Pass Filter and 10dB pad Includes AT8449 SV shockmount

MICROPHONE PREAMPS

AVALON DESIGN VT-737SP Mono Class A. Vacuum Tube-Discrete Preamp-Opto-Compressor-Equalizer



The VT-737SP is a vacuum tube. Class A processor that combines a mic preamp, instrument DI, compressor and sweepable 4-band equalizer in a 2U rack space. Like all Avalon Design products the VT-737SP utilizes a minimum signal path design with 100% discrete, high-bias pure Class A audio amplifiers and the best active and passive components available. Used by renowned artists and studios world wide and the winner of the Electronic Musiciar 1999 Editors' Choice Award for Product Of The Year.

FEATURES-

- ombination of TUBE preamplifiers, opto-compressor sweep equalizer, output level and VU metering in a 2U space
- Four dual triode vacuum tubes, high-voltage discrete Class A with a 10 Hz to 120kHz frequency response
- ±0.5dB . The Preamp has three input selections- The first is a high performance XLR balanced mic input transformer
- with +48v phantom power, the second is a high impedance instrument DI with a 1/4" lack located on the front panel and the third is a discrete high-level Class A
- balanced line input. · High gain switch boosts overall preamp gain and a passive- variable high pass filter, hardwire relay bypass and phase reverse relay is available for all
- The Opto-Compressor uses a minimum signal path design and features twin Class A vacuum tube triodes for gain matching. A passive optical attenuator serves as a simple level controller. Variable threshold, compression ratio and attack and release offer dynamics control from soft compression to hardknee limiting.
- The dual sweep mid-EQ can be side chained to the compressor allowing a broad range of spectra

control including de-essing. The EQ can be assigned pre and post compressor from the front panel to add

- even greater sonic possibilities. Two VT-737 SPs can be linked together via a rear panel link cable for stereo tracking • The Equalizer utilizes 100% discrete, Class A-high-
- voltage transistors for optimum sonic performance. The low frequency passive shelving EQ is selectable
- between 15, 30 60 and 150HZ with a boost and cut of ±24dB
- The high frequency passive shelving EQ is selectable between 10, 15, 20 and 32 kHZ with a buost and cut of ±20dB . The low-mid frequency is variable between 35 to 450
- Hz while the high-mid frequency is variable from 220Hz to 2.8 kHz. Both mid-band frequencies offer a boost and cut of ± 16 dB and a hi-Q/lo-Q switch.
- When the EQ to side chain is used, the low and high EQ is still available for tonal adjustment
- The Output level is continuously variable and utilizes an another dual triode vacuum tube driving a 100% Class A, high-current balanced and DC coupled low noise output amplifier
- · Sealed silver relay bypass switches are used for the most direct signal path

POWERED STUDIO MONITORS

VERGENCE A-20



- -6dB LE Cutoff 40Hz
- 5 position wall proximity control
- 5 position listening proximity control between near. mid and far-field monitoring • Power, Overload; SPL Output, Line VAC and Dutput
- device temperature display.

Speakers

- 2-way acoustic suspension with a 6.5-inch treated paper woofer and a 1-inch aluminum dome tweeter
- · Fully magnetically Shielded with an 18-inch
- recommended working distance



The PS-5s are small format, full-range, non-fatiguing project studio monitors that give you the same precise, accurate sound as the highly acclaimed 20/20 series studio monitors. The use of custom driver components, complimentary crossover and bi-amplified power design

provides a wide dynamic range with excellent transient response and FLECTRONCS low intermodulation distortion.

FEATURES

420 Ninth Ave.

(Bet. 33rd & 34th St.)

New York, N.Y. 10001

neutral studio reference monitoring system for

A-20's control amplifier adapts to any production

input sensitivity while the speakers magnetic

48Hz - 20kHz frequency response @ 1M

XLB outputs from power amp to speakers

· Matched impedance output cables included

computer based studios

(100ms peak). • XLR, TRS input connectors

1M)

Amplifier

shielding allows seamless integration into today's

Type Modular, self-powered near/mid/far-field monitor

· Peak Acoustic Output 117dB SPL (100ms pink noise at

· Amplifier Power 250W (continuous rms/ch), 400W

Headphone output
 5-position input sensitivity switch with settings

- 5-1/4-inch magnetically shielded mineral-filled polypropylene cone with 1-inch diameter high-temperature voice coil and damped rubber surround LF Driver
- Magnetically shielded 25mm diameter ferrofluid-cooled natural silk dome neodymium HF Driver · 70 watt continuous LF and 30 watt continuous HF
- amplification per side XLR-balanced and 1/4-inch (balanced or unbalanced)

Also Available- TRM-8

1-inch soft dome tweeter and 8-inch

polypropylene woofer • 45Hz • 21kHz frequency response ±2dB

. 75 Watt HF, 150 Watt LF amplification

Offering honest, consistent sound from top to bottom, the TRM-6 bi-amplified studio monitors are the ideal reference monitors for any recording environment whether tracking, mixing and mastering. Supported by Hafler's legendary amplifie technology providing a more accurate sound field, in width, height and also depth

FFATURES-

- 33 Watt HF & 50 Watt LF amplification
- 1-inch soft dome tweeter and 6.5-inch polypropylene woofer
- 55Hz 21kHz Response
- · Magnetically Shielded · Electronically and Acoustically Matched

TRM-10s And TRM-12s **Active Subwoofers**

Combining Hafler's legendary amplifier technology with a proprietary woofer design, the TRM10s and TRM12s active subwoofers provide superb bass definition required in today's studio and surround sound environments

TRM-10s

- 10-inch cellulose fibre cone down firing woofer.
- 200 watt low frequency amplifier
 30Hz to 110Hz frequency response ±2dB
- · 24dB/octave Linkwitz-Riley crossover variable (40Hz to 110Hz)

TRM-12s

- 12-inch cellulose fibre cone down firing woofer
 200 watt low frequency amplifier
- 25Hz to 110Hz frequency response ±2dB
 24dB/octave Linkwitz-Riley crossover variable (40Hz to
- 110Hz)







- crossover Built-in RF interference,
- Combination Power On/Clip LED indicator
 5.8 vinyl-laminated MDF cabinet

TRM-6 Bi-Amplified Studio Monitors











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taken in complete measure. As an idling studio,

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Database Editor: Management position to coordinate the most extensive database development project in the history of the MI industry. Individual must be able to coordinate research, solicitation, and compilation of extensive database information on products, contacts, documents, etc.



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TC WORKS

ULTIMATE SOFTWARE MABILINES



TC Native Bundle*

TC Electronic hardware effects processors are the cornerstone of many top recording studios. These five plug-ins, featuring the incredible TC native reverb, bring that legendary TC-quality audio



processing to your MOTU system desktop. The TC Native Bundle neatly places at your fingertips over 20 years of audio processing R&D, deployed in native 32-bit floating point glory. Incredible sound, well-crafted presets, low CPU overhead, and intuitive controls make these TC Native plug-ins a joy to use.



ChannelStrip"

ChannelStrip is like having a magebuck mixing console inside your PowerMac. Even artists who regularly use top-of-theline, large format consoles are raving about the "high-end console" sound they get from ChannelStrip. How did Metric Halo do it? By combining 61 standard, fully-automatable audio processing facilities into a single, complete plug-in with 64-bit floating point precision. ChannelStrip is heavily optimized for efficient operation in your MOTU native recording environment, so you can use it throughout your mix. How does ChannelStrip actually sound? Producer Andy Gray-Ling puts it like this: "...I'm absolutely mindblown. It sounds amazing..."

ANTARES



COMPATIBLE

Antares Microphone Modeler"

Now the microphones you own can sound like the microphones you wished you owned. Mic Modeler allows any reasonably full-range microphone to sound like virtually any other mic. Using patented Spectral Shaping Tool" (SST) technology, Antares has created precise digital models of a wide variety of microphones, from historical classics to modern exotics, as well as a selection of industry-standard workhorses. Just select which microphone you are actually using and then select what mic you want it to sound like. You can further fine-tune the sound with modeled tube saturation, proximity, windscreen effect, and more. Mic Modeler is an easy, cost effective way to extend your existing mic collection, or to obtain that classic, vintage sound without the excessive price tag.



LOUD



RealVerb[®]

On the heels of their groundbreaking RealVerb 5.1[™] sourround reverb plug-in, Kind of Loud Technologies presents RealVerb[™], a new stereo reverb plug-in for MAS. RealVerb uses complex spatial and spectral reverberation technology to accurately model an acoustic space. The bottom line? Great sounding reverb with the ability to customize a virtual room and pan within the stereo spectrum. RealVerb even lets you blend room shape, material, and size according to the demands of your mix. And RealVerb was designed from the ground up for automation: adjust controls in real-time without distortion, pops, clicks or zipper noise. You can even morph between presets – in real-time. Don't rely on your old standby – let RealVerb bring new quality and space to your recordings.

2408mkll audio interface"

To mix your project with these advanced plug-ins, listen to it through our new 2408mkII audio interface — now with balanced quarter-inch, 24-bit analog I/O

(8 in / 8 out), with inputs that are switchable between +4/-10, plus a volume knob for the main outs. Same price. Same incredible product. Just more value.

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17-41

CIRCLE 52 ON FREE INFO CARD





Surround sound speaker suggestions and how to get over on over lights

BY ROGER NICHOLS

I'm late with my column this month, so I tried to use an updated excuse for my tardiness. In high school, the favorite excuse was always, "My dog ate my homework." Or the one I used was, "I put it on your desk as I was leaving yesterday. You didn't get it? Gee, that was my only copy."

With the advent of computers, the "that was my only copy" ploy no longer works. But "my dog ate my computer" seems like a good one.

Nope! The editor isn't going for it. He said if I didn't turn it in by the end of today, he was going to flunk me...so here goes.

SURROUND STUFF

As I am writing this, Elliot Scheiner and I are printing the last of the surround mixes for a Steely Dan DVD that will be out this summer. What I think is different about this DVD is that the focus is primarily on the quality of the music portion with the video taking a back seat. The DVD will have a stereo PCM channel, a Dolby Digital AC-3 sur-

round channel, and a DTS surround channel. If there is any room left on the disc, it will be filled with video.

The speaker setup for the surround mixes was as follows: The front right and front left speaker were 6 feet, 6 inches apart (tweeter to tweeter) and 6 feet, 6 inches from the mix position. The center front speaker was halfway between these two, but still 6 feet, 6 inches from the mix position, which made it a little farther back from the console bridge. Oh, the left and right speakers were on stands behind the console, not sitting on the meter bridge.

The rear speakers were a mirror image of the front. Six feet, 6 inches apart, 6 feet, 6 inches from the mix position. Herein lies the rub. This is the accepted positioning for the nearfield monitors when mixing 5.1 surround, but I don't know anyone with that setup for listening at home.

The front speaker setup is fine. It doesn't

matter to me if the playback speakers are a little higher, or a little farther apart, just as long as the stereo spread sounds reasonable. The rear speakers are a different thing altogether.

If you are facing forward while mixing in this configuration, you can get a pretty good balance of all the instruments in all five speakers, but, if you turn your head just a little either way,

Over lights can not really measure anything that is over. because you can never get any signal louder than all of the bits used up.

the rear speaker information seems too loud. I think this is because the rear speakers are too close together and the sound just hits the back of your head, or the back of your ears, and you don't get much sonic information. If you move the rear speakers farther apart or if you

move them up higher, the definition improves tremendously. This wider rear-speaker positioning also scems to be the preferred setup in most homes with surround systems, including mine. I have noticed that, at home, I have to turn down the rear-speaker level for most 5.1 surround music discs, as compared to the levels I set for surround movie soundtracks. Film soundtracks are mixed in a different acoustic environment with the speaker positions more closely matching those in a theater setting. The rear speakers are higher and farther apart.

So what does all this have to do with the price of corn? Nothing. But I think that the standard speaker setup for surround mixing should be modified so that the end-user doesn't have to change the level of his rear speakers to hear the proper balances on music surround product. So there!

IT'S OVER BETWEEN US

I get a lot of questions about over lights on DAT machines, Pro Tools meters, and digital processor

boxes. Here is everything you ever wanted to know about overs, but were afraid to ask.

Over lights can not really measure anything that is over, because you can never get any signal louder than all of the bits used up.

The over lights have to guess at whether the signal would have gone continued on page 132



This Mic Is Anything But Flat...



...That's Why The Pros Love It.

The Neumann M 147 Tube

For years, vintage Neumann tube mics, such as the venerable U 47, have been high-priced, highly prized commodities. Why, when advances have created mics with near-perfect, virtually transparent reproduction, have producers and engineers travelled to the ends of the earth in search of these vintage relics? Because of the way they sound (especially the way the sound sits in the mix).

Enter the M 147 Tube.

Using the same capsule as the classic U 47 and its smaller cousin the U 47 FET, the M 147 Tube microphone brings a warmth, presence and detail to vocals that is simply unattainable from any other mic being produced today, regardless of how much it looks like a Neumann. The fact is, there is really only one way to get that classic sound you seek. Fortunately, it's priced well within your reach.



Neumann – The Choice of Those Who Can Hear The Difference

What The Professionals Are Saying About The Neumann M 147 Tube:

"So far, I'm thrilled to pieces with the Neumann M 147 Tube. I don't think there's any instrument that I wouldn't try them on. Whatever instrument I used them for, I was very impressed with the sound. I wish I had about five or six of them!"

- Al Schmitt, as quoted in EQ, March 1999

"I would recommend the M 147 highly for rock, rap, pop, jazz or blues vocals; drum room and/or kick drum miking; all tube and solid-state instrument amplifiers; nylon string guitar; and low-volume or indistinct sound sources that need some extra presence. and for any type of digital recording. In short, I like the M 147 a lot -- so much so that I bought one," - Myles Boisen,

Electronic Musician, August 1999

"The particular kind of presence it adds is really unique and desirable, and it's really not available from any other mic or easily obtainable with an equalizer. Typically, condenser mics that have a forward character are really just brittle and edgy, and the M 147 is completely different from that.' - Monte McGuire.

Recording, July 1999

"I asked the singer on my session which mic she preferred and, when presented with a finite budget, her pick (and mine) was the M 147. Classic Neumann sound, tube electronics, the U 47 legacy, and a price that won't savage your bank account. Gotta love it!" - Rick Chinn, **Audio Media**

February 1999

"The M 147 proves again that however close the imitators get, there is no substitute for the genuine article. This is the real McCoy and although it cannot be called cheap, its simple approach means that it is far more accessible than a valve Neumann would normally be expected to be. Another classic in the making." - Dave Foister,

Studio Sound. February 1999 "It's my opinion that the tone of the Neumann would not require much EQing during mixdown; a decided advantage. Its high end would sit nicely in a mix, and its round but controlled low end would not have to be cut to provide room for other instruments." - Mitch Gallagher,

Keyboard Magazine, lune 1999

CIRCLE 39 ON FREE INFO CARD

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- +4 balanced XLRs
- AES/EBU with Sample Rate Conversion
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The new high-end audio workstation.

- The MOTU 1296 a 24-bit / 96 kHz audio workstation for Macintosh or Windows, available as a core system and expansion I/O.
- Impeccable analog I/O 12 independent inputs and outputs on +4 balanced XLRs.
- Stunning sound 24-bit / 96 kHz converters with 117 dB dynamic range (A-weighted @ 48 kHz); extremely low-jitter crystal.
- Advanced design isolated analog circuit & R/CORE transformers.
- Flexible AES/EBU digital I/O sample rate converters on input and output, with independent crystal and word clock input.

 Ideal for surround mixing and recording — supports two 5.1 surround mixes in and out simultaneously.

- Expandable connect up to three 1296 interfaces to a single computer for 36 channels of 96K input and output.
- · Compatible with your favorite audio software includes standard ASIO 2.0 and Wave drivers for Lacin
- Upgrade your current MOTU system recording to your 2408, 1224 or any



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